TOMORROW
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Cisco ISR G2 2900/3900 and Cisco ISR 4451-X: Unified Communications on Cisco ISR routers

BRKUCC-2260

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Agenda

- Introduction
- Review of ISR G2 voice gateways and recent changes to the portfolio
- ISR 4451-X as a voice gateway
- What's new in Cisco Unified Communications Manager Express
- What's new in SRST Manager/ E-SRST
- UC applications on Cisco UCS E-series
- Analog voice gateways
- Conclusion

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Unified Communications on the Cisco 2900/3900/3900E ISR Generation 2 (G2)

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Integrated Services Routers Generation 2

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Cisco 2900 Series

Integrated Services Routers

	2951	2921	2911	2901
SM Slots	2	1	1	0
ISM Slots	1	1	1	1
EHWIC Slots	4	4	4	4
Onboard DSP Slots	3	3	2	2
Onboard WAN Ports	3 GE (1 SFP)	3 GE (1 SFP)	3 GE	2 GE
Default Flash	256 MB	256 MB	256 MB	256 MB
Default DRAM	512 MB	512 MB	512 MB	512 MB
Form Factor	2RU	2RU	2RU	1RU

Secure Collaboration Platform

- Up to 75Mbps WAN Access with Services
- Increased service density with Second Services module Slot
- 12 Inch Depth on 2911

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Cisco 3900 Series

Integrated Services Routers

	3945E	3925E	3945	3925
Field Upgradeable Motherboards	SPE-250	SPE-200	SPE-150	SPE-100
Onboard WAN	4GE (2 SFP)	4GE (2 SFP)	3GE (2 SFP)	3GE (2 SFP)
Perf w/ Services	350 Mbps	250 Mbps	150 Mbps	100 Mbps
SM Slots	4	2	4	2
ISM Slots	0	0	1	1
EHWIC Slots	3	3	4	4
Onboard DSP Slots	3	3	4	4
Opt Dual PS	Yes	Yes	Yes	Yes
Default Flash	256MB	256MB	256MB	256MB
Default DRAM	1 GB	1 GB	1 GB	1 GB
Form Factor	3RU	3RU	3RU	3RU

Scalable Rich-media Services Platform

New Services Performance Engine 200s:

- Up To 350Mbps WAN Access With Services
- 4 GE ports onboard

All 3900 Series:

- Up to 1040 watts with PoE Boost
- Configurable dual Integrated Redundant Power supplies
- Hot swappable fans and power supply

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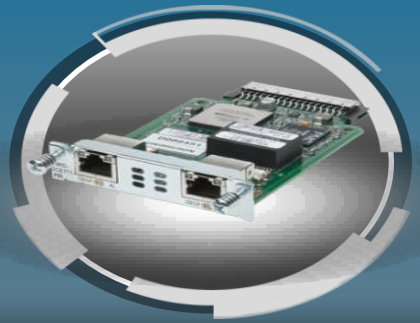
Service Modules and Interface Cards

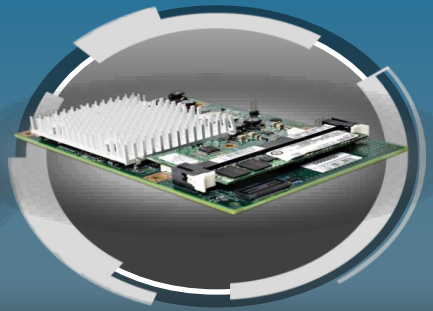
Interface Cards (WAN or LAN)



EHWIC
Enhanced High Speed WAN Interface Card

Internal Module for Running Services using router ports

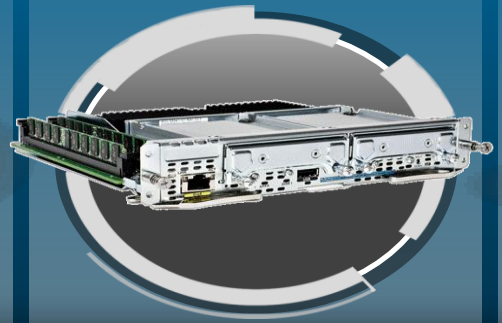
Example: Cisco Unity Express



ISM
Internal Service Module

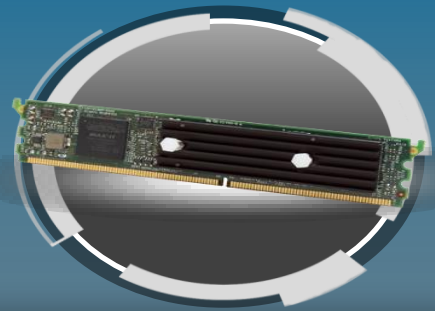
Hosting Services with external interface ports.

Examples: Wireless LAN Controller, WAN Optimization, Etherswitch Module



SM
Service Module

High Density Rich-Media Voice and Video DSP Modules



PVDM3
Packet Voice/Video DSP Module

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UC Module Support Summary

EHWIC Slots **VIC Cards**	• VIC cards are not supported • VIC2 cards are supported unless superseded by a VIC3 card • All VIC3 cards are supported

SM Slots
NM, EVM Cards
	• Existing EVM supported using SM-NM-adapter card
	• EVM-based EM cards are supported
	  • NM-HDA-specific EM cards are not supported
	  • EM3-HDA-8FXS/DID supported (old EM-HDA-8FXS is not)
	• No dedicated EVM slots, all SM slots can be used
	• SM-SRE servers and UCS E-series for UC collaboration applications

ISM Slots **AIM/ISM Cards**	• No AIM form factor cards are supported • ISM-SRE-300-K9 supported for CUE
DSP Slots **PVDM2, PVDM3**	• Motherboard DSP slots use PVDM3s • PVDM2s supported using PVDM2-adapter card

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UC Module Support on the ISR G2s

Supported
Analog and BRI Cards (VIC)
VIC2-2FXO
VIC2-4FXO
VIC2-2BRI-NT/TE
VIC3-2E/M
VIC3-2FXS/DID
VIC3-2FXS-E/DID
VIC3-4FXS/DID

Supported
Digital T1/E1 Cards (VWIC)
VWIC2-1MFT-T1/E1
VWIC2-2MFT-T1/E1
VWIC2-1MFT-G703
VWIC2-2MFT-G703
VWIC3-1MFT-T1/E1
VWIC3-2MFT-T1/E1
VWIC3-1MFT-G703
VWIC3-2MFT-G703
HWIC-1CE1T1-PRI
HWIC-2CE1T1-PRI
VWIC3-4MFT-T1/E1
EVM and EM Cards
EVM-HD-8FXS/DID[1]
EM-4BRI-NT/TE
EM-HDA-3FXS/4FXO
EM-HDA-6FXO
EM3-HDA-8FXS/DID

[1] Requires SM-NM-ADPTR card

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UC Module Support on the ISR G2s

Legacy UC modules	Reached end-of-sale
Network Modules (NM)	
NM-HD-1V	✓
NM-HD-2V	✓
NM-HD-2VE	✓
NM-HDV2	✓
NM-HDV2-1T1/E1	✓
NM-HDV2-2T1/E1	✓
Application Modules	
NME-CUE	✓
NME-UMG	✓
NME-UMG-EC	✓
NM-CUSP-522	✓
NME-APPRE	✓
ISM-SRE-300-K9 (CUE)	
SM-SRE-700-K9 (CUE)	

Legacy DSPs	Reached end-of-sale
DSP Cards (DSP, PVDM, EC)	
PVDM2-8	✓
PVDM2-16	✓
PVDM2-32	✓
PVDM2-48	✓
PVDM2-64	✓
EC-MFT-32	✓
EC-MFT-64	✓
PVDM3-16	
PVDM3-32	
PVDM3-64	
PVDM3-128	
PVDM3-192	
PVDM3-256	
Adapter Cards	
PVDM2-ADPTR	
SM-NM-ADPTR	

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ISR T1/E1 Interface cards support and scale

- Only HWIC slots available for VWIC cards
- Single T1/E1 clock domain on ISR G2
- Three T1/E1 clock domains on ISR 4451-X
- ISR G2 limited to 16 T1/E1 ports
- ISR 4451-X supports up to 24 T1/E1 ports and will support up to 40 in future

ISR G2 modules
VWIC3-1MFT-T1/E1
VWIC3-2MFT-T1/E1
VWIC3-1MFT-G703
VWIC3-2MFT-G703
VWIC3-4MFT-T1/E1

ISR 4451-X modules
NIM-1MFT-T1/E1
NIM-2MFT-T1/E1
NIM-4MFT-T1/E1
NIM-8MFT-T1/E1
NIM-1CE1T1-PRI
NIM-2CE1T1-PRI
NIM-8CE1T1-PRI

Platform	T1/E1 Ports		
	2-port Card	4-port Card	8-port Card
2901	8	N/A	N/A
2911, 2921	8	16	N/A
2951, 3925	8	16	N/A
3945	8	16	N/A
3925E	6	12	N/A
3945E	6	12	N/A
4451	6	12	24

Cisco NM-HDV2 modules have reached End-of-Sale

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TDM GW Overall Positioning

Platform	Channels		T1s		E1s	
	2-port VWIC	4-port VWIC	2-port VWIC	4-port VWIC	2-port VWIC	4-port VWIC
2901	100	100	4	N/A	3	N/A
2911	150	150	6	6	5	5
2921	240	240	8	10	8	8
2951	240	400	8	16	8	13
3925	240	480	8	16	8	16
3945	240	480	8	16	8	16
3925E	180	360	6	12	6	12
3945E	180	360	6	12	6	12

Platform	Channels			T1s			E1s		
	2-port NIM	4-port NIM	8-port NIM	2-port NIM	4-port NIM	8-port NIM	2-port NIM	4-port NIM	8-port NIM
4451	180	100	720	6	12	24	6	12	24

End-of-sale of NM-HDV modules limits T1/E1 ports on ISR G2

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TDM Gateway Analog and BRI ports

VIC (Voice Interface cards)	2901	2911	2921	2951	3925	3945	3925E	3945E
FXS	16	40	40	64	64	112	60	108
FXO/CAMA	16	28	28	40	40	64	36	60
E&M	8	12	12	16	16	24	14	22
Analog-DID	16	32	32	48	48	80	44	76
BRI Ports	8	16	16	24	24	40	22	38
BRI Channels	16	32	32	48	48	80	44	76

NIM (Network Interface Module)	4451
FXS	12
FXO/CAMA	12
E&M	12
Analog-DID	12
BRI Ports	12

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Overall 2900/3900 UC Positioning

Platform	Data Circuit Speed	TDM Channels		CUBE Sessions	CTS Sessions		CME Phones	SRST Phones
		2-port VWIC	4-port VWIC		1000	3000		
2901	25M	100	100	100*	-	-	35	35
2911	35M	150	150	200*	2	1	50	50
2921	50M	240	240	400*	4	2	100	100
2951	75M	240	400	600*	6	3	150	250
3925	100M	240	480	800*	10	5	250	730
3945	150M	240	480	950*	17	10	350	1200
3925E	250M	180	360	2000*	28	17	400	1350
3945E	350M	180	360	2500*	40	24	450	1500

*Performance based on basic call setup

Limited due to EoS of NM-HDV

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Unified Communication on ISR 4451-X

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ISR 4400 Platform Introduction

NG Branch Platforms

10Gb

5Gb

2Gb

1Gb

500Mb

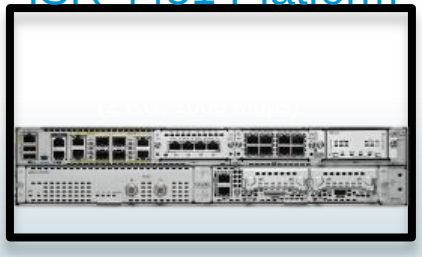
350Mb

250Mb

ASR1002
(2.5G – 5G)

ASR1001
(1.25G – 2.5G)

ISR 4451 Platform



Launched
June 2013

3945E

3925E

* ISR 4451 platforms add AVC and embedded next-gen services to ISR G2 services mix.

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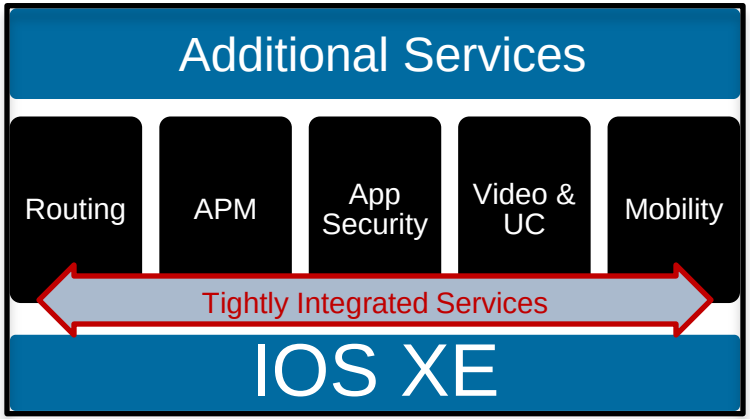
ISR 4451
Architecture Overview



ISR 4451 Platform

What is the ISR 4451?

An Enterprise Gateway that securely delivers video, voice, and business applications from private or public cloud to remote users with a rich user experience.

What does it offer?

Offers best-in-class and integrated application network services (Layer 2 through Layer 7 **services** and **applications**).

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ISR 4451 Front View

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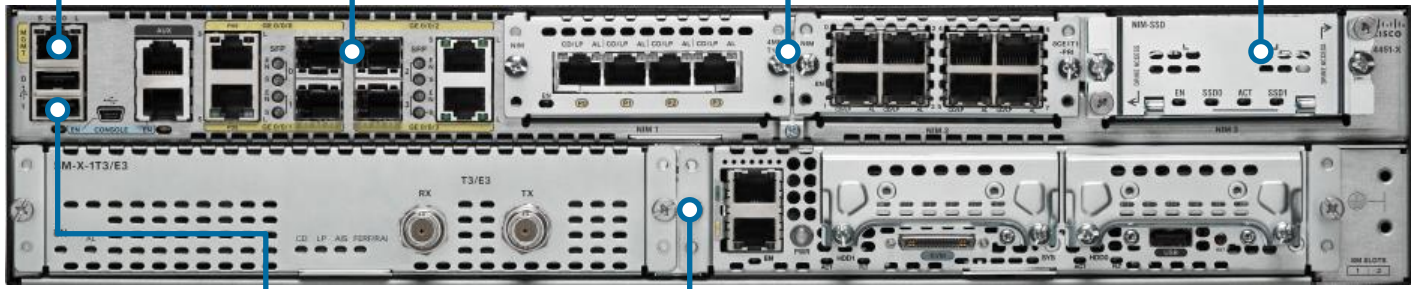
ISR 4451 IO Design

Management Interface
Connects router's control plane directly to a management network.

Front Panel GE
- 4 RJ45/SFP GE Interfaces
- PoE available on 2 Interfaces

Network Interface Modules (NIM)
- Larger & more powerful than EHWICs
- Up to 8 ports per module
- DSPs directly on modules

Dual Drive Slots for Applications
- RAID 1 for data protection
- HD & SSD



USB Connections
- 2 x Type A for file storage and charging your cell phone
- USB Console in addition to RJ45 Console and Aux ports

Extended Service Modules
- Compatible with ISR G2
- Up to 10Gb connection to system
- Faster & more powerful than SMs

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Next Generation Branch Platform

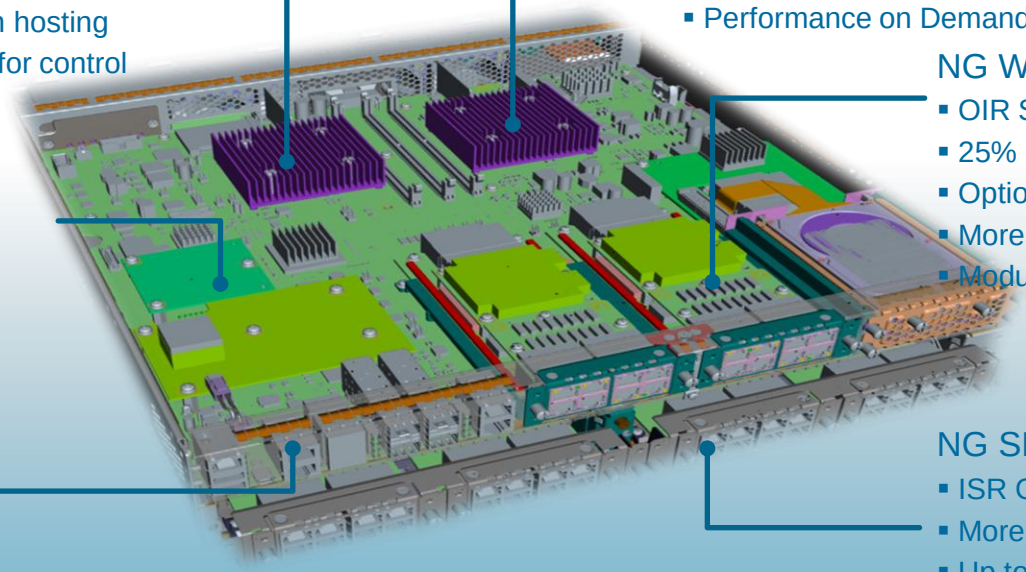
Under the Covers

Dedicated Control Plane CPUs
- On-board application hosting
- Stable performance for control plane functions
- Modular OS

Next Generation DSP
- Better Density for voice
- More compact form-factor

Multi-Core Data Plane
- Scalable Performance
- Performance on Demand

NG WIC Modules
- OIR Support
- 25% better performance
- Option dual-HD for applications
- More real-estate for interfaces
- Modular driver architecture



Front-Panel Interfaces
- 1 Management GE
- 4 SFP+RJ45 GE
- PoE Capable

NG SM Modules
- ISR G2 Compatible
- More power and cooling
- Up to 10x better performance

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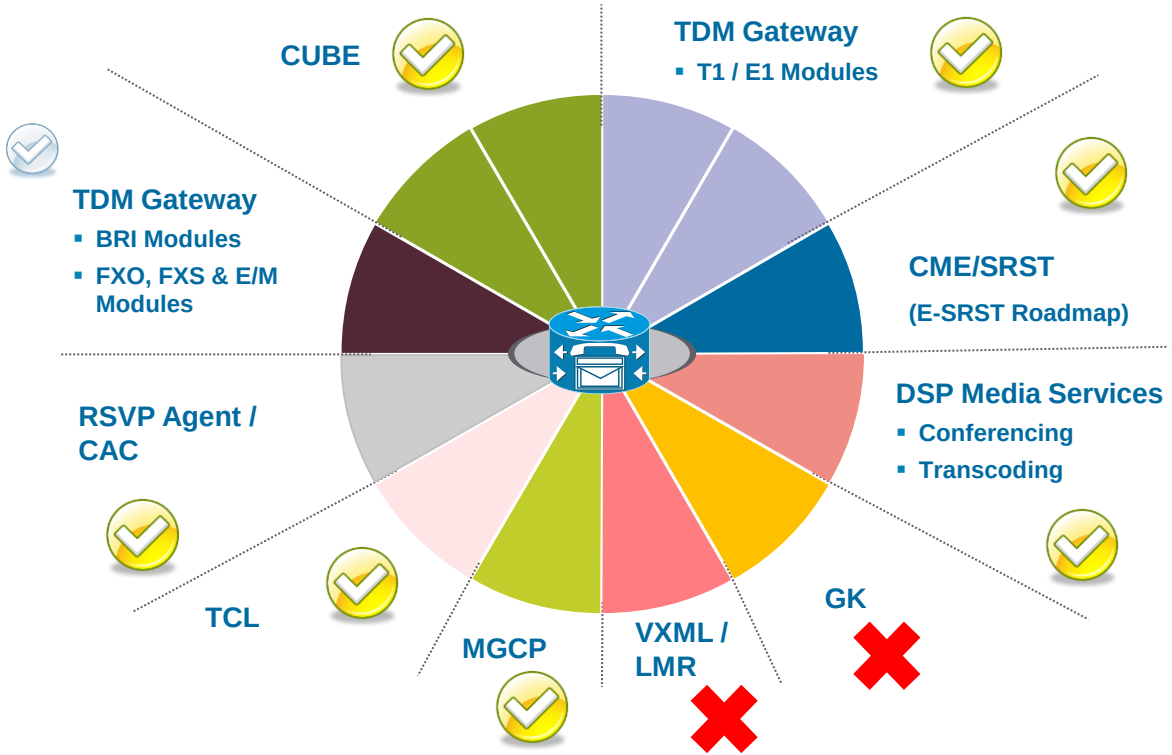
Unified Communications on ISR 4451-X

UC on ISR 4451 – Initial Offering

Dec 2013

Aug/Sep 2014



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Unified Communications on the ISR 4451

Cisco IOS XE 3.11 / 3.12

- TDM Functionality
  - T1/E1 CAS/PRI
- MGCP
  - T1 CAS
  - MGCP PRI
- Cisco Unified Border Element
  - CUBE Enterprise
  - Dialplan enhancements
- SIP & H.323
- Cisco Unified Communications Manager Express
- Cisco Survivable Remote Site Telephony
- Digital Signal Processor – PVDM4

To be delivered XE 3.13

- BRI
- Analog Interfaces – FXS/FXO/E&M

Roadmap

- E-SRST
- Gatekeeper
- High-density DSP farm
- LMR
- E1 R2

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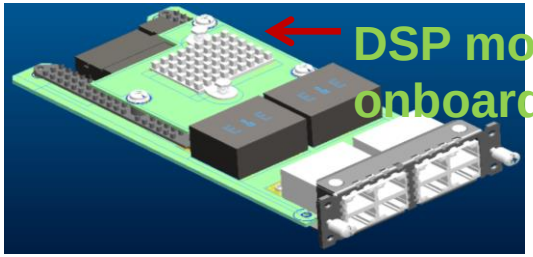
T1/E1 and DSP Overview

- 4th Generation Voice and WAN NIM
- 1, 2, 4 and 8 Port T1/E1 module
- Onboard NIM DSP module for TDM services
- NEBS Level 3 compliant
- Up to 720 E1/576 T1 channels
- Support for T1 MGCP
- IOS trunk-group feature



	NIM Slots	SM Slots	Onboard DSP Slots
ISR 4451	3	2	1

2-port T1/E1 NIM module

PVDM4 (DSP) module

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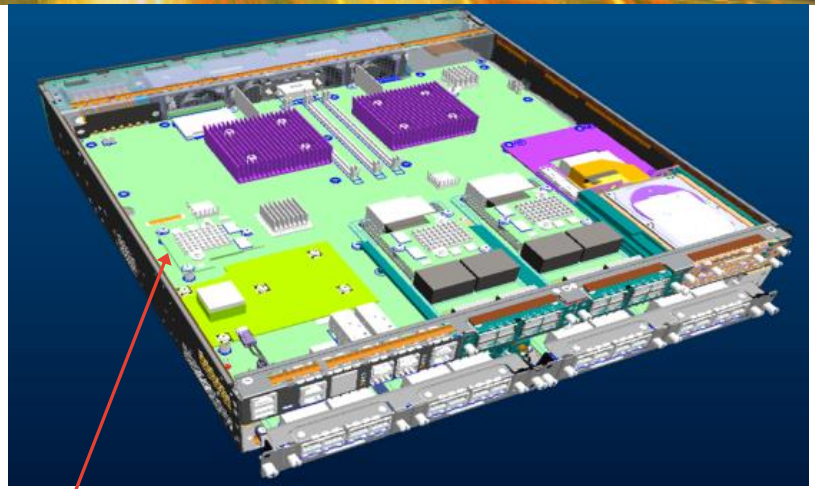
Cisco PVDM4

- DSP module for TDM and IP services
- 4 PVDM4 SKUs
- One PVDM4 can be installed on router mother board for IP services (transcoding, conference and MTP)
- One PVDM4 can be installed on each NIM module for TDM and IP services
- A maximum of 4 PVDM4 modules can be installed using NIM
- Voice conference bridge, MTP and transcoding supported (video conferencing is not supported)



PVDM4-32	1 DSP module with 32 G.711 channels
PVDM4-64	1 DSP module with 64 G.711 channels
PVDM4-128	1 DSP module with 128 G.711 channels
PVDM4-256	1 DSP module with 256 G.711 channels

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TDM Channels

- ISR 4451 can be fully populated with TDM interfaces and run at full capacity.
- Table below shows the TDM channels supported on ISR 4451
- One 8-port T1/E1 NIM can handle 192 calls
- One PVDM4-256 can support 192 simultaneous G729a calls (medium complexity)

Platform	Max Channels				Max T1/E1 ports
	1-Port NIM	2-Port NIM	4-Port NIM	8-Port NIM	
ISR 4451	90	180	360	720	24

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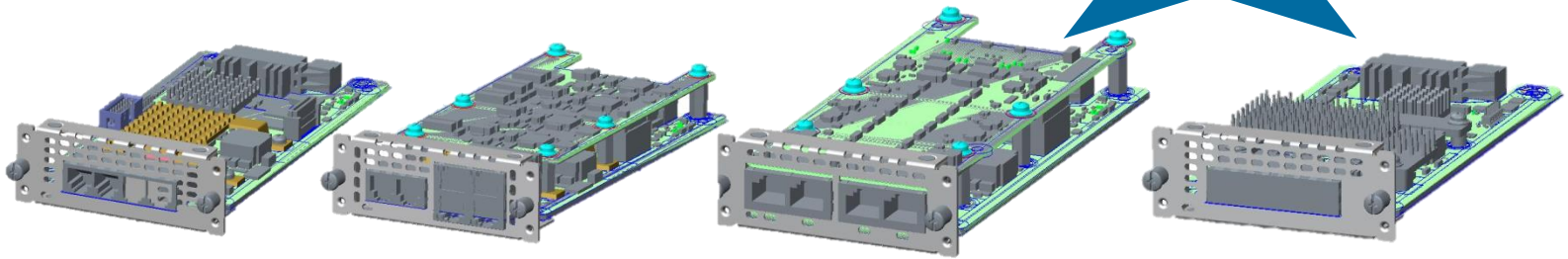
Cisco 4451 Analog NIM voice cards

- Cisco NIM analog voice cards have built-in DSP and do not require PVDM4
- All FXS NIM cards support FXS-Extended loop length (18,000ft using 24AWG wire)
- Voice traffic does not impact GE throughput capability of 4451
- Available NIM voice cards:
  - NIM-2FXS, NIM-4FXS, NIM-2FXO, NIM-4FXO, NIM-2FXS/4FXO
  - NIM-4E/M, NIM-2BRI-NT/TE, NIM-4BRI-NT/TE

New, available in August 14'

New, available in Sep14'



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Gateway Signaling Protocol Summary

Analog TDM

- FXS
- FXO
- E&M: wink, immediate, delay dial
- Analog DID
- CAMA

Digital TDM

- BRI: Q.931, QSIG
- T1: CAS, FGD, PRI, QSIG, PRI NFAS
- E1: PRI, QSIG, R2, CAS/MELCAS
- T3

VoIP

- SIP
- H.323
- MGCP 1.0: SP call agents
- MGCP: 0.1+: Unified CM

- Signaling protocol and interface support varies across the GW platforms
- No voice T3 NIM card available for ISR G2 or 4451-X
- A subset of TDM protocols is supported with MGCP and varies depending on the call agent used with the GW
- SS7 is not supported on ISR G2 or 4451-X

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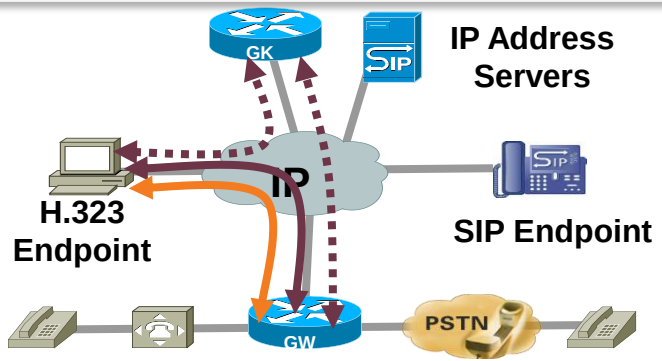
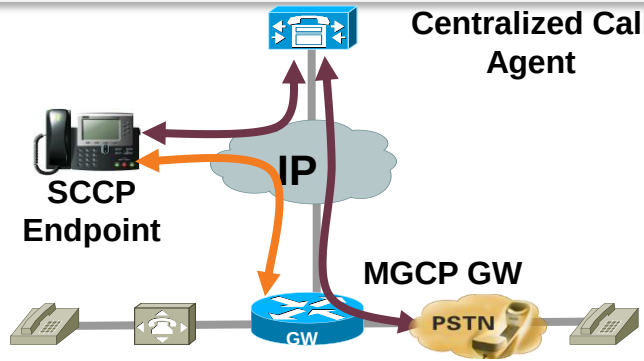
Call Control Models and Protocols

Distributed Call Control
Peer-to-Peer Protocols: SIP, H.323



- Peer-to-peer call setup—IP address servers optional
- GW TDM signaling types local to GW
- Resilient over IP connectivity failures
- Scalable
- Distributed configuration

Centralized Call Control
Client-Server Protocols: SCCP, MGCP



- Call agent arbitrates call setup between endpoints
- GW TDM signaling types dependent on call agent
- Dependent on IP connectivity between endpoint and call agent
- SRST or ESRST required for resiliency
- Centralized configuration

Optional Signaling to Locate IP Address of Peer
Call Control Signaling
Media (RTP)

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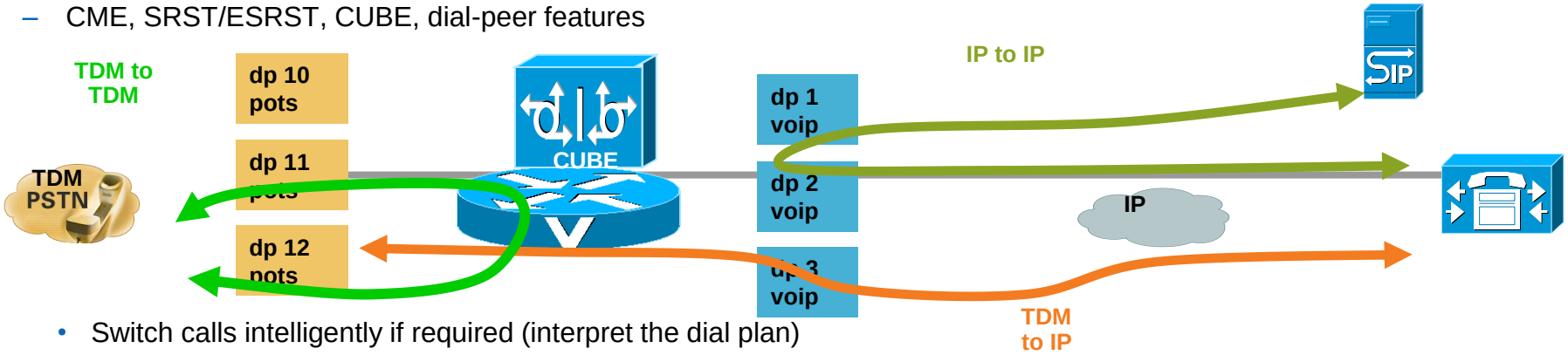
Voice on Cisco 4451-X

- Cisco 4451-X voice services will be feature parity with ISR G2
  - CME, SRST/ESRST, CUBE, dial-peer features



- Switch calls intelligently if required (interpret the dial plan)
- Digit manipulation (called, calling and numbering plan)
- Failover (preferences) to alternate destinations
- Load balancing
- Video/Data ISDN switching and termination
- Integrated voice, video and data PRIs
- Insert applications into the call path: Tcl/VXML

  Build support for signaling variations (e.g. CLID on T1 CAS)

  Hookflash trunk release on FXO

  VXML call control for call centers

  Redistribute calls-in-q for CVP

  AA in the GW

These Capabilities do Not Exist for MGCP-Controlled GWs

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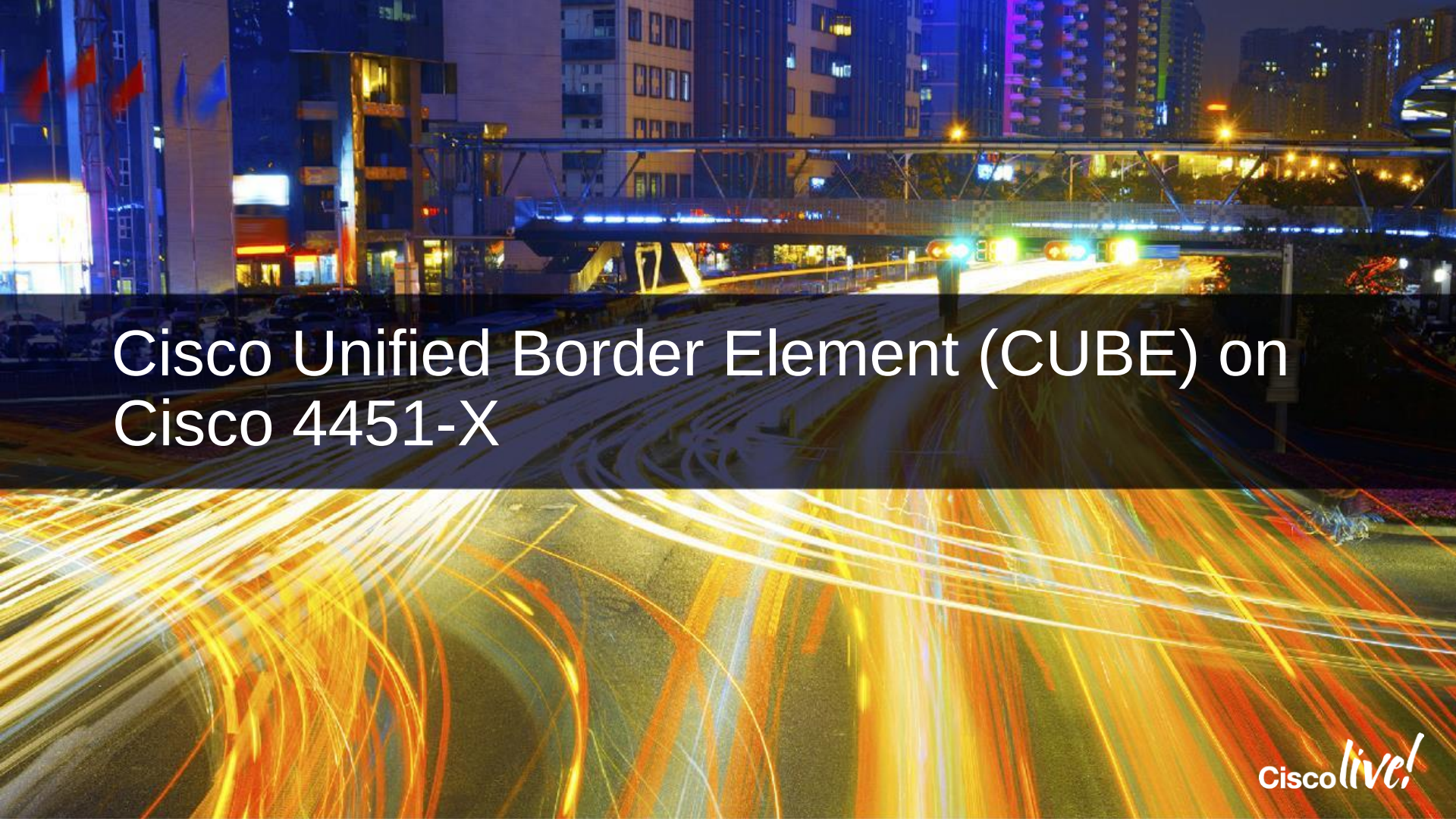

Cisco Unified Border Element (CUBE) on Cisco 4451-X

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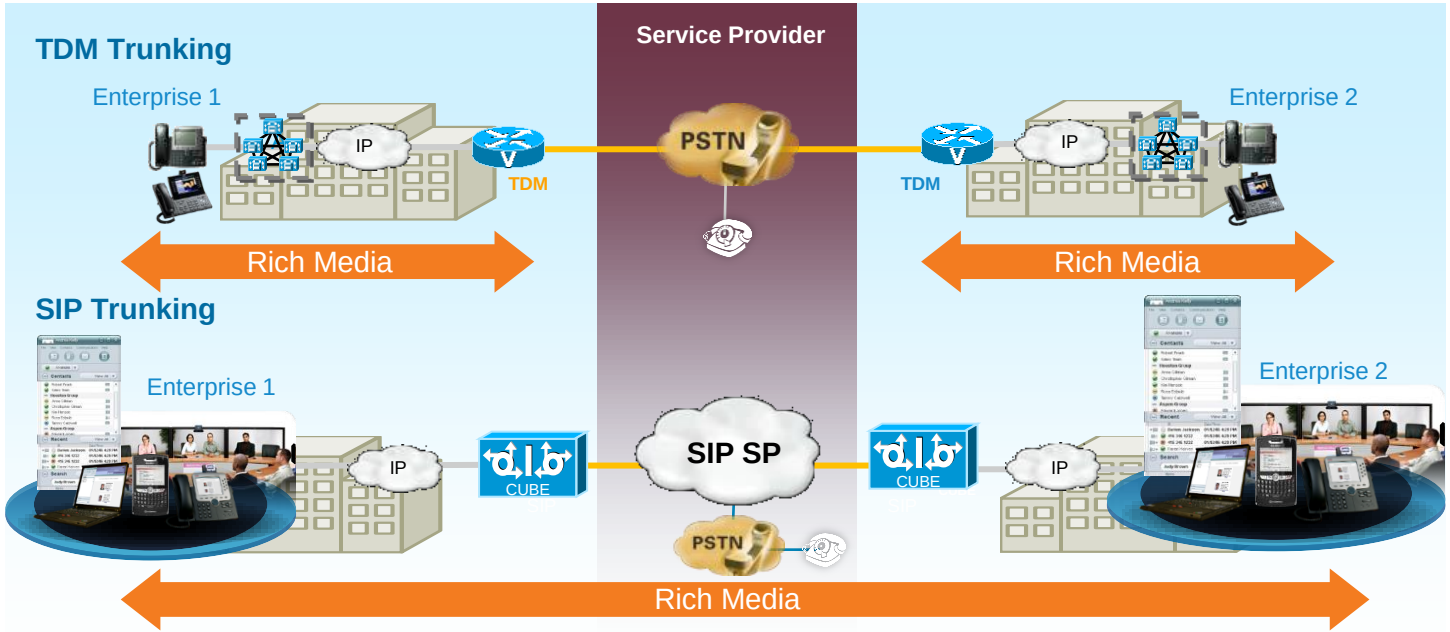
SIP Trunking Overcomes TDM Barriers

- Improves Efficiency of interconnection between networks
  - Simplifies PSTN interconnection with IP end-to-end
- Enables rich media services to employees, customers, partners
  - Carries converged voice, video and data traffic



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Why does an enterprise need an SBC ?

SESSION CONTROL

Call Admissions Control
Ensuring QoS
Statistics and Billing
Redundancy/ Scalability

SECURITY

Encryption
Authentication
Registration
SIP Protection
Firewall Placement
Toll Fraud

INTERWORKING

SIP - SIP
H.323 - SIP
SIP Normalization
DTMF Interworking
Transcoding
Codec Filtering

DEMARCATION

Fault Isolation
Topology Hiding
Network Borders
L5/L7 Protocol
Demarcation

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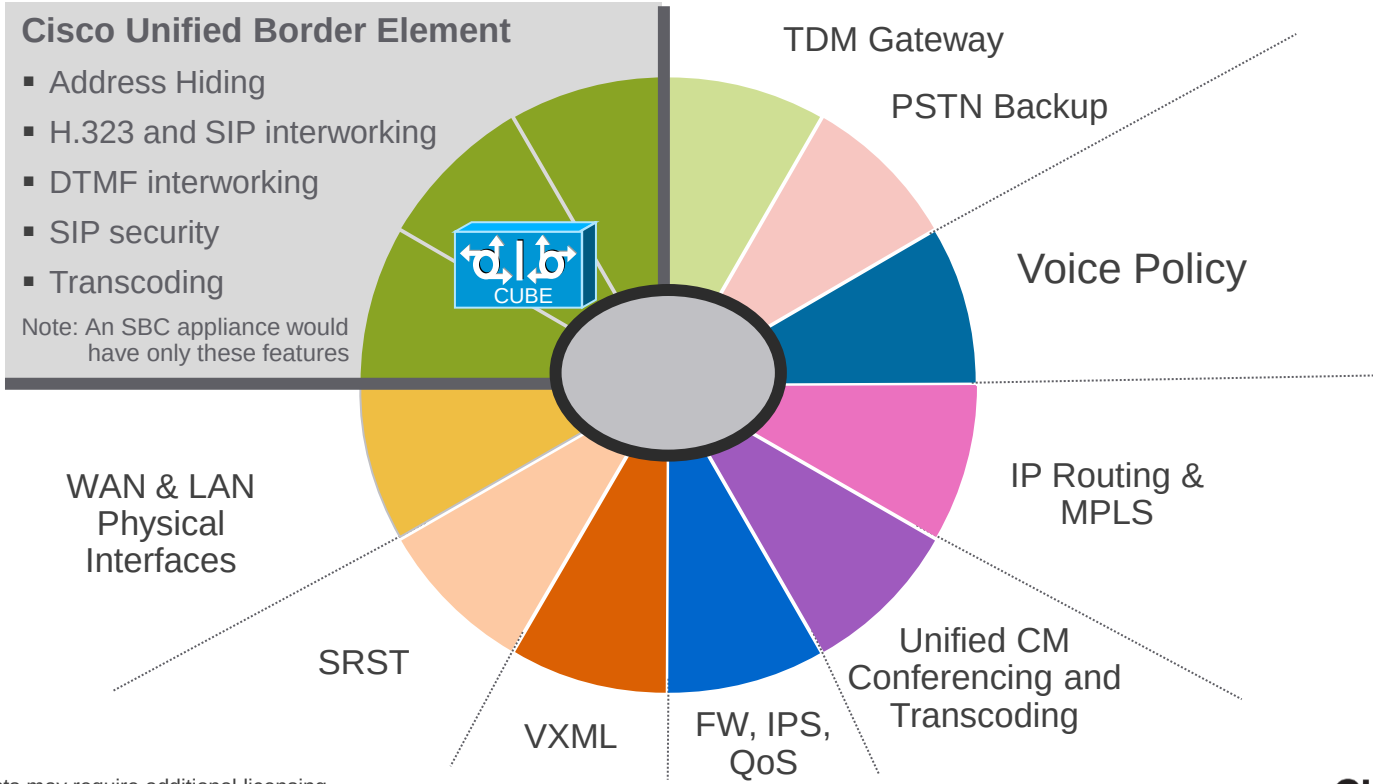
Cisco Unified Border Element – More Than an SBC

An Integrated Network Infrastructure Service



Note: Some features/components may require additional licensing

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CUBE (Enterprise grade SBC)

- Scales up to 16000 sessions (basic calls on ASR1006 platform)
- SBC platform portfolio to meet the right size for your enterprise
- New features for dial plan management to support high scale SIP trunk integration

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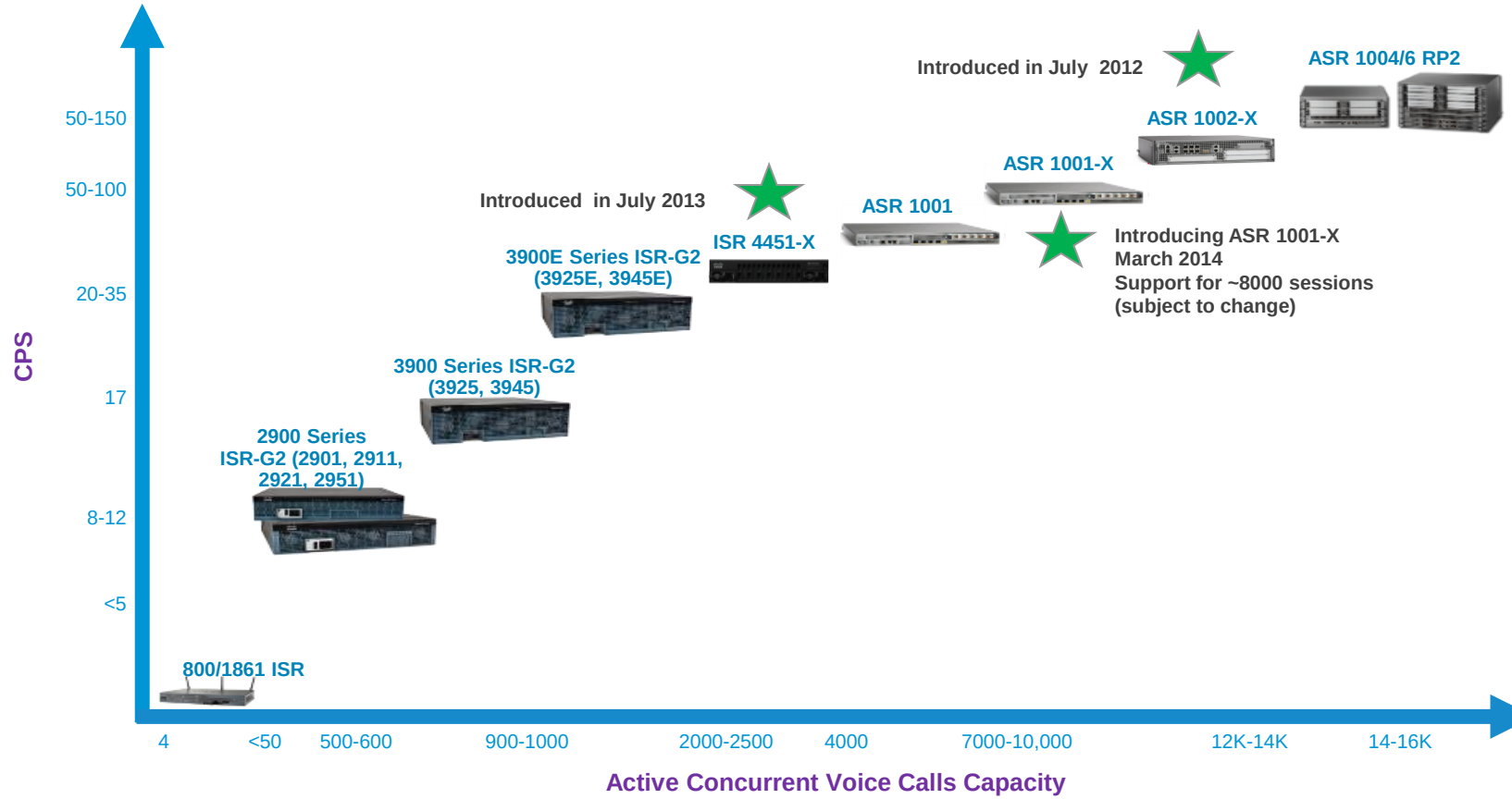
CUBE (Enterprise) Product Portfolio



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Cisco Public

Dialplan enhancement teaser
To learn more please meet with Husain Ali
on Thursday at 2:30PM BRKUCC-2934
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Destination Server Group

15.4.1T/ XE3.11

- Supports multiple destinations (session targets) be defined in a group and applied to a single outbound dial-peer
- Once an outbound dial-peer is selected to route an outgoing call, multiple destinations within a server group will be sorted in either round robin or preference [**default**] order
- This reduces the need to configure multiple dial-peers with the same capabilities but different destinations. E.g. Multiple subscribers in a cluster

```
voice class server-group 1
 hunt-scheme {preference | round-robin}
 ipv4 1.1.1.1 preference 5
 ipv4 2.2.2.2
 ipv4 3.3.3.3 port 3333 preference 3
 ipv6 2010:AB8:0:2::1 port 2323 preference 3
 ipv6 2010:AB8:0:2::2 port 2222

* DNS target not supported  in server group
```

```
dial-peer voice 100 voip
 description Outbound DP
 destination-pattern 1234
 session protocol sipv2
 codec g711ulaw
 dtmf-relay rtp-nte
 session server-group 1
```


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Multiple Destination-Patterns Under Same Outbound Dial-Peer

15.4.2M/ XE3.7

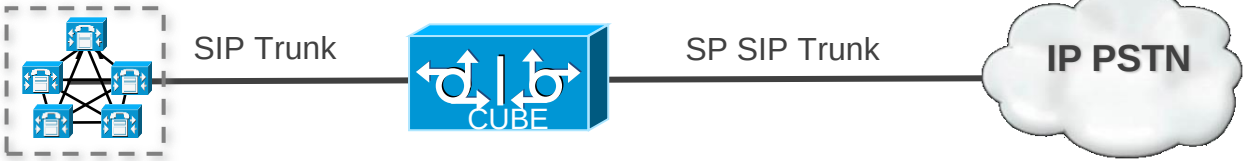
Site A (919)200-2000
Site B (510)100-1000
Site C (408)100-1000

G729 Sites

```
voice class e164-pattern-map 100
 e164 919200200.
 e164 510100100.
 e164 408100100.

dial-peer voice 1 voip
 destination e164-pattern-map 100
 codec g729r8
 session target ipv4:10.1.1.1
```

Provides the ability to combine multiple destination-patterns targeted to the same destination to be grouped into a single dial-peer



Site A (919)200-2010
Site B (510)100-1010
Site C (408)100-1010

G711 Sites

```
voice class e164-pattern-map 100
 url flash:e164-pattern-map.cfg

dial-peer voice 1 voip
 destination e164-pattern-map 100
 codec g711ulaw
 session target ipv4:10.1.1.1
```

```
! This is an example of the contents
  of E164 patterns text file
  stored in flash:e164-pattern-
  map.cfg

9192002010
5101001010
4081001010
```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Multiple Incoming Patterns Under Same Incoming Dial-peer

15.4.1T/ XE3.11

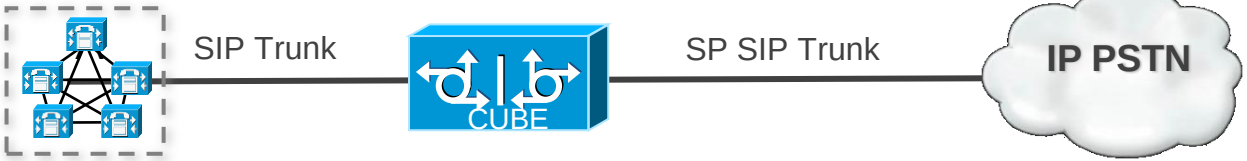
Site A (919)200-2000
Site B (510)100-1000
Site C (408)100-1000

G729 Sites

```
voice class e164-pattern-map 100
 e164 919200200.
 e164 510100100.
 e164 408100100.

dial-peer voice 1 voip
 description Inbound DP via Calling
 incoming calling e164-pattern-map 100
 codec g729r8
```

Provides the ability to combine multiple incoming called OR calling numbers on a single inbound voip dial-peer, reducing the total number of inbound voip dial-peers required with the same routing capability



Site A (919)200-2010
Site B (510)100-1010
Site C (408)100-1010

G711 Sites

```
voice class e164-pattern-map 200
 url flash:e164-pattern-map.cfg

dial-peer voice 2 voip
 description Inbound DP via Called
 incoming called e164-pattern-map 200
 codec g711ulaw
```

```
! This is an example of the contents
  of E164 patterns text file
  stored in flash:e164-pattern-
  map.cfg

9192002010
5101001010
4081001010
```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Cisco Unified Border Element

- Split plane architecture allows increased performance on the ISR 4451
- Table below lists the supported sessions on an ISR 4451

Platform	Release	CUBE Sessions Flow Through	CUBE Sessions Flow Around
ISR 4451-X	XE 3.11	4000	10000

Please attend CUBE session BRKUCC-2934 with Hussain Ali
Thursday 2:30PM to 4pm at Moscone West 3001

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Cisco Unified Communications Manager Express (CME)

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Cisco Unified Communications Manager Express

Simple, Affordable All-In-One Communications

Now available on Cisco 4451-X running XE3.12

- Unified Communications solution for 450 users or less based on Cisco Integrated Services Routers
- Full-featured solution that meets the needs of distributed enterprise branch office, and service-provider-managed-service
- Provides robust networking across sites using H.323 or SIP
- Voice mail, auto-attendant and IVR capabilities with integrated Cisco Unity Express
- Configure as PBX, Key Switch or hybrid mode for business continuity
- Options include: Video Telephony, Tele-workers, Cisco IP Communicator soft phone, B-ACD call routing, SIP phone support, plus integration with Cisco Unified Communications Manager
- Provisioning and fault reporting using Cisco Configuration Professional and Cisco Unified Operations Manager and Arcana ManageExpress iManage

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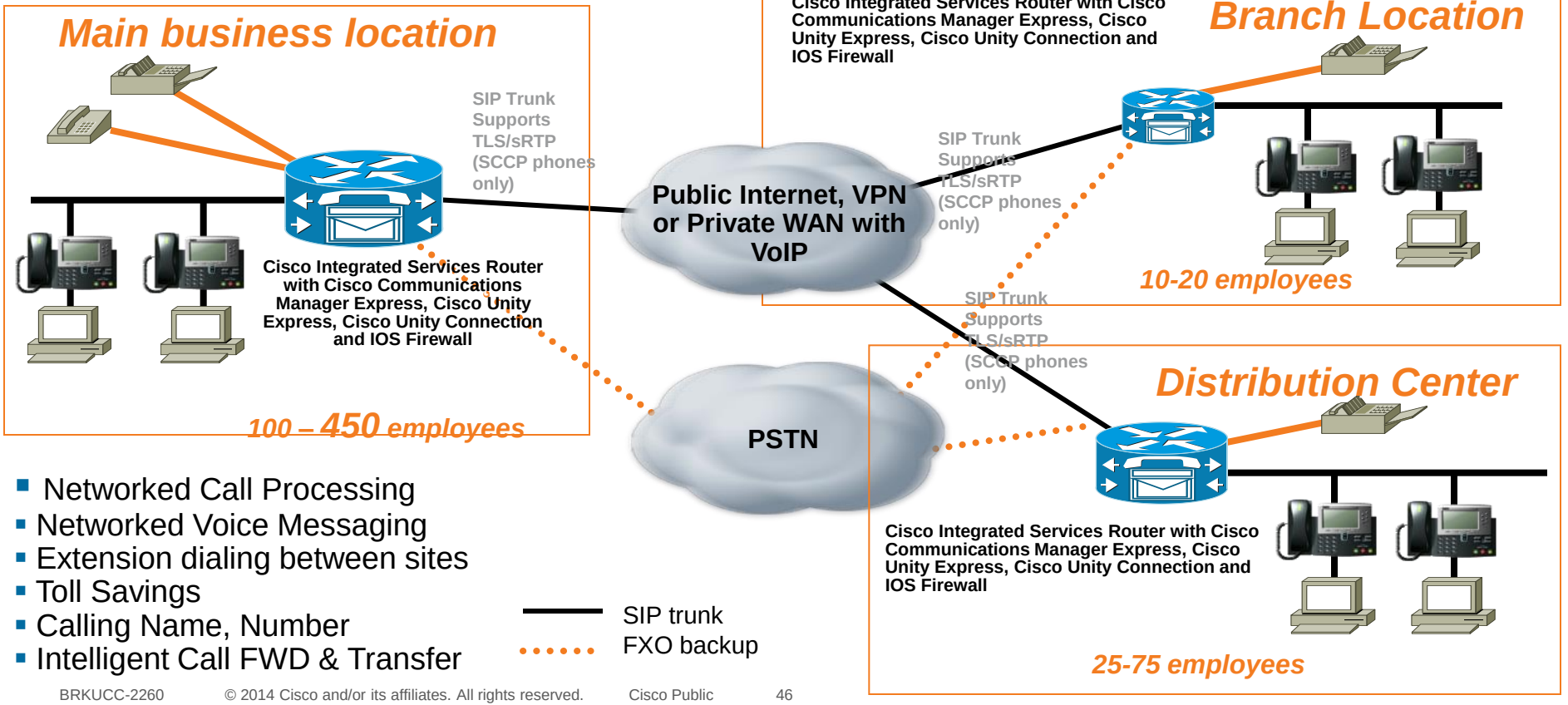
Unified Communications Express: Distributed Enterprise Branch Office


Main business location
SIP Trunk Supports TLS/sRTP (SCCP phones only)
Cisco Integrated Services Router with Cisco Communications Manager Express, Cisco Unity Express, Cisco Unity Connection and IOS Firewall
100 – 450 employees
Public Internet, VPN or Private WAN with VoIP
PSTN
Cisco Integrated Services Router with Cisco Communications Manager Express, Cisco Unity Express, Cisco Unity Connection and IOS Firewall
Branch Location
SIP Trunk Supports TLS/sRTP (SCCP phones only)
10-20 employees
SIP Trunk Supports TLS/sRTP (SCCP phones only)
Distribution Center
Cisco Integrated Services Router with Cisco Communications Manager Express, Cisco Unity Express, Cisco Unity Connection and IOS Firewall
25-75 employees
SIP trunk
FXO backup

- Networked Call Processing
- Networked Voice Messaging
- Extension dialing between sites
- Toll Savings
- Calling Name, Number
- Intelligent Call FWD & Transfer

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Cisco Unified Communications Manager Express

What's New?

Coming June release

- **Cisco IP phones 7800 and 8800 SIP**
  Using Fast-Track CME now supports latest IP phones
- **Dynamic Hunt-group login**
  A single DN can log in and out of multiple hunt-groups
- **Visual list of Parked calls**
  List of all parked calls with option to select and retrieve
- **CME SIP multi VRF support**
  CME will support SIP IP phones registered through VRF (match CME SCCP VRF feature support)

Latest CME features and admin guide can be found at:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmeroad.html

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CME support for Cisco 7800 and 8800 IP phones

- Using CME Fast-Track feature a ‘pool-type’ for each non-natively supported end-point can be configured and mapped to a supported reference IP phone type
- When CME development adds native support of a new IP phone to a later CME release, the new CME code only removes the ‘fast-track config portion of the CME cli and keeps the voice register pool configuration with native “type XX” command
- Using fast-track also allows modifying the xml-config file of the new phone type in order to support added KEM module as an example

xml-config is used to define the phone specific XML tags to be used in configuration file.

Examples: xml-config maxNumCalls 3 xml-config busyTrigger 3

where maxNumCalls defines the maximum number of calls allowed per line

busyTrigger defines the number of calls that triggers call forward busy per line on the sip phone.

Note that CME would populate these values in the CNF file and the same would be downloaded to the phone

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Cisco CME and Cisco 7800 IP phones

Fast-track configuration

New phone model	Recommended Fast-track config	
7821	voice register pool-type 7821 xml-config maxNumCalls 6 xml-config busyTrigger 6 xml-config custom sshAccess 0 num-lines 2 description Cisco IP Phone 7821 reference-pooltype 6921	voice register pool 11 id mac 6C41.6A36.85FE session-transport tcp type 7821 (must match pool-type label) number 1 dn 11 username 1121 password 1121 codec g711ulaw
7841	voice register pool-type 7841 xml-config maxNumCalls 6 xml-config busyTrigger 6 xml-config custom sshAccess 0 num-lines 4 description Cisco IP Phone 7841 reference-pooltype 6941	voice register pool 12 id mac 6C41.6A36.85FF session-transport tcp type 7841 number 1 dn 12 username 1122 password 1122 codec g711ulaw
7861	voice register pool-type 7861 xml-config maxNumCalls 6 xml-config busyTrigger 6 xml-config custom sshAccess 0 num-lines 16 description Cisco IP Phone 7861 reference-pooltype 6961	voice register pool 13 id mac 6C41.6A36.8600 session-transport tcp type 78261 number 1 dn 13 username 1123 password 1123 codec g711ulaw

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Cisco CME and Cisco 8800 IP phones

New phone model	Recommended Fast-track config	New phone model
8841	voice register pool-type 8841 xml-config maxNumCalls 6 xml-config busyTrigger 2 num-lines 5 description Cisco IP Phone 8841 reference-pooltype 8961	voice register pool 21 id mac A436.7234.CCE1 session-transport tcp type 8841 number 1 dn 21 username 2121 password 2121 codec g711ulaw
8851	voice register pool-type 8851 xml-config maxNumCalls 6 xml-config busyTrigger 2 addons 2 num-lines 77 description Cisco IP Phone 8851 reference-pooltype 9951	voice register pool 22 id mac A436.7234.CCE2 session-transport tcp type 8851 number 1 dn 22 username 2122 password 2122 codec g711ulaw
8861	voice register pool-type 8861 xml-config maxNumCalls 6 xml-config busyTrigger 2 addons 3 num-lines 113 description Cisco IP Phone 8861 reference-pooltype 9971	voice register pool 23 id mac A436.7234.CCE3 session-transport tcp type 8861 number 1 dn 23 username 2123 password 2123 codec g711ulaw
8831	voice register pool-type 8831 xml-config maxNumCalls 4 xml-config busyTrigger 4 xml-config custom sshAccess 0 description CISCO IP Phone 8831 reference-pooltype 6921	voice register pool 24 id mac A436.7234.CCE4 session-transport tcp type 8831 number 1 dn 24 username 2124 password 2124 codec g711ulaw

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Cisco CME Dynamic hunt-group login

Feature detail:

- Any single DN (SIP/SCCP) can dynamically join/unjoin multiple voice hunt groups. A DN can join dynamically to a maximum of 6 hunt-groups
- Using the service button “My-Phone-Apps” a user can display all available voice hunt-groups and obtain status of which hunt-groups the user is currently joined
- To join or unjoin a voice hunt-group a user can use either FAC code or Softkey
- PLK button support to access the voice hunt group menu on the SIP/SCCP phone
- Custom FAC aliasing is supported only for SCCP phones. Not for SIP phones
- On SCCP phones, Line and phone level login/logout (HLog) to voice hunt groups is supported and on SIP phones, only line level login/logout is supported

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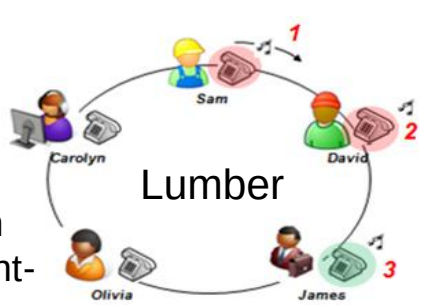
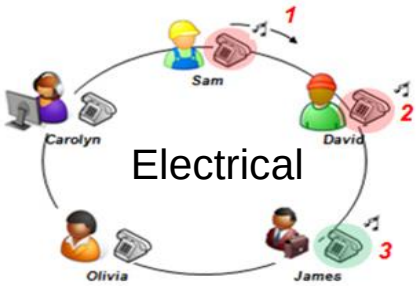
Configuring Cisco CME Dynamic hunt-group login

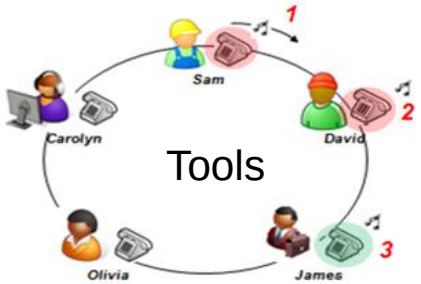
voice hunt-group config:

Voice hunt-group 1 parallel
List 1111, 2222, 3333, *, *, *, *
Phone-display

- The *(star) character represents the dynamic login slot, there are 4 dynamic slots assigned to this hunt-group
- The “Phone-display” cli is required in order to push hunt-group status information to IP phone display







DN config:

For SCCP phones:
ephone-dn 1
voice-hunt-groups login

For SIP phones:
voice register dn 1
voice-hunt-groups login

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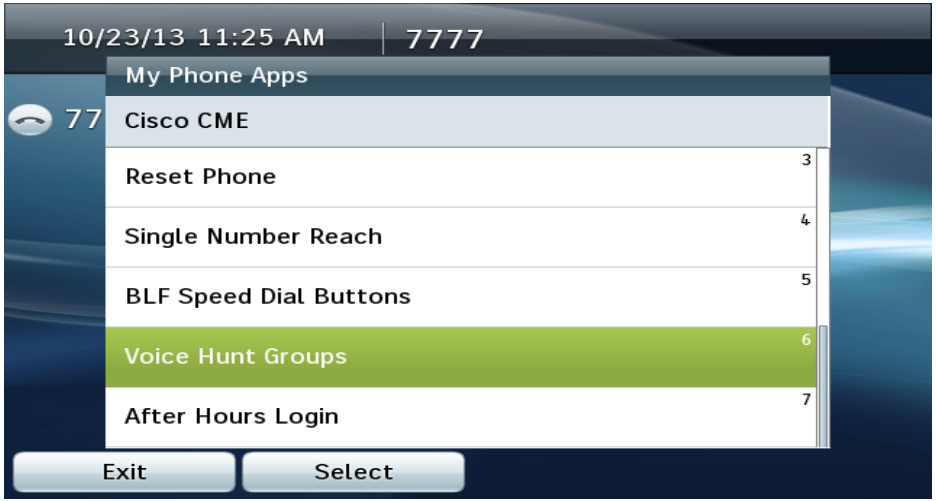
Configuring Cisco CME Dynamic hunt-group login

Standard FAC:
Telephony-service
Fac standard (If the" fac" standard is configured the user must use the standard FAC for SIP and SCCP phones)
ephone-hunt join *3 ---------- *3 + followed by voice hunt group pilot number
ephone-hunt cancel #3--------To unjoin from all the voice hunt groups.
ephone-hunt hlog *4---------- For line level Hlog
ephone-hunt hlog-phone *5----For phone level HLog.
ephone-hunt unjoin #4------unjoining from particular voice hunt group

Custom FAC:
Telephony-service
Fac custom fac custom ephone-hunt
If the FAC custom is configured, the user has to use the following with their own custom FAC standards.

Cancel	Select all hunt-group join cancel feature
Hlog	Select hunt-group hlog feature
hlog-phone	Select hunt-group hlog phone feature
join	Select hunt-group join feature—Unjoin from particular voice hunt.
Unjoin	Select hunt-group unjoin feature

For SIP phones to make use FAC's:
Telephony-service
call-park system application



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Configuring Cisco CME Dynamic hunt-group continued

PLK button configuration for voice hunt groups info display on phone:

For SCCP phones:

ephone-template 1
url-button 1 voicehuntgroups ➔ configure this under ephone template to enable PLK button
.
ephone 10
ephone-template 1 ➔ Assign the ephone-template created in above step

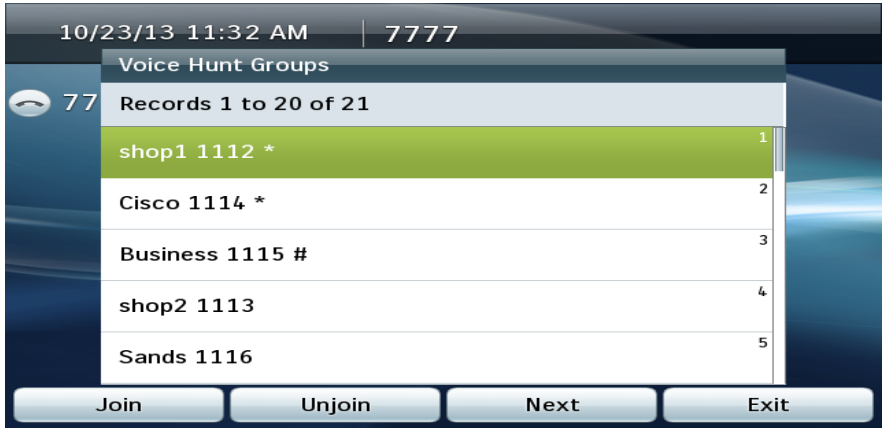
For SIP phones:

voice register template 1
url-button 3 http://x.x.x.x:80/CMEserverForPhone/vhg_root_menu Label ➔
Where "**x.x.x.x"** is the CME source address and "**label"** is any name you assign.

voice register pool 1
template 1 ➔ Assign the voice register template created in above step

To enable or disable voice hunt-group menu options under "My phone Apps" feature use the following commands:
Ephone 1
phone-ui voice-hunt-groups

Ephone 1
no Phone-ui voice-hunt-groups



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Cisco CME Visual list of parked calls

Feature detail:

- Current CME version does not display any parked call related information
- Status monitoring of parked call could only be done via cli. A user parking calls would have to remember which call was parked in which slot and how long has it been parked
- With the new “visual list of parked calls” feature a user can call up a list to display on the Cisco IP phone to visually monitor parked calls and retrieve a parked call if desired
- All listed parked calls also display the length of time the call has been parked for
- The active parked calls will be displayed on the phone via menu option via “My Phone Apps” and “Park List” service button. The parked call list feature can be directly launched using PLK, if configured.

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Cisco CME Visual list of parked calls configuration

The following information related to all active parked call will be displayed on the phone

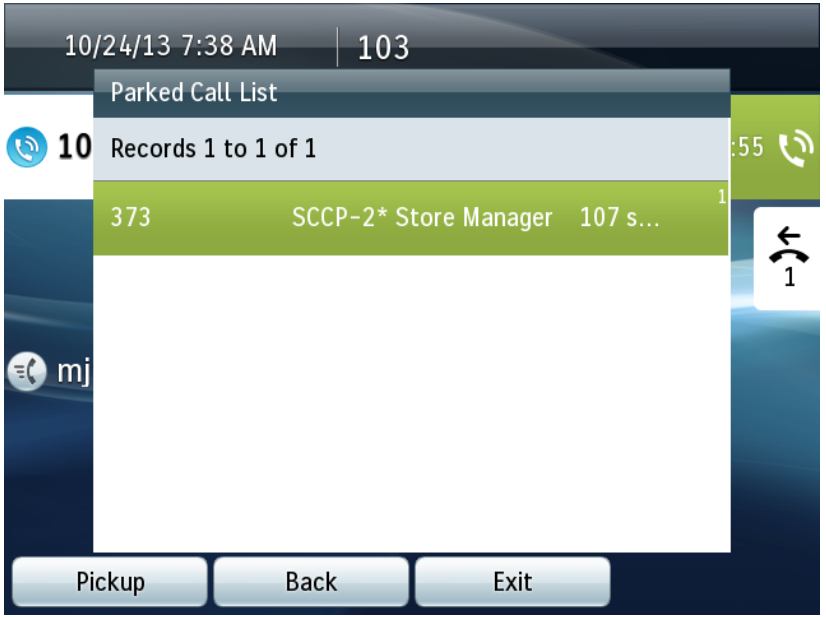
- Park slot name (number if name is not available)
- Parker name (number if name is not available)
- External name (number if name is not available)
- Age of the call (in seconds)

To enable Park-list feature under my phone apps, add following config under ephone.

phone-ui [park-list]

To enable the feature in PLK, add following config under ephone.

url-button 1 [park-list]



Limitations, Restriction and Dependencies

- This feature for “call-park application system” only
- Only supported for “ephone-dn” park slot
- Requires *ip http* or *ip https* services running on the router
- The list feature supports up to 20 parked calls
- Reserved parked slots are not supported

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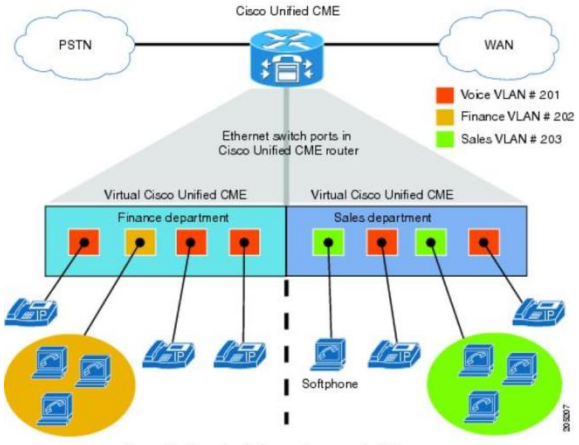
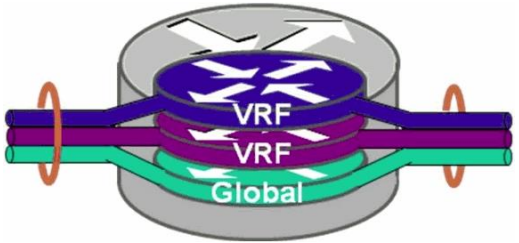
Cisco CME SIP VRF support

Feature detail:

- Virtual Routing and Forwarding (VRF) is an IP technology that allows multiple instances of a routing table to coexist on the same router at the same time
- Only one global CME call control is supported on a router with multiple VRF
- Current CME versions do not support SIP end-points to register from different VRFs. All SIP devices must come in to register from a single global VRF, this is in contrast to SCCP IP phones
- With the new feature CME will support SIP end-point registration and media flow from multiple VRFs





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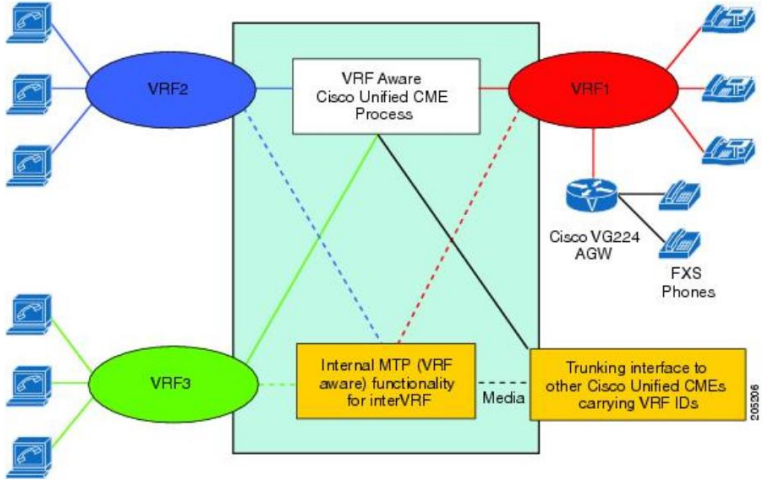
Cisco CME SIP VRF

- VRF is used to virtualize L3 network domains using the same network hardware
- One use case is separating data user groups into separate L3 networks, but allowing voice communication between the separate user groups without compromising data separation
- Another use case CME can be virtually partitioned into multiple Cisco Unified CME routers.
  - Example: a single Cisco Unified CME being shared by both a supermarket and a Coffee shop/Bank in a campus, but as networks, they can not talk to each other except over a PSTN or through restricted route leaks



Keep in mind

- CME must run on the “voice VRF”, the voice vrf must be explicitly configured
- Dial-peers are only supported on voice VRF
- DSP resources (xcoding, conferencing, MTP) register through the voice vrf

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Cisco CME SIP VRF configuration

Create VRFs

```
vrf definition data-vrf1
 rd 102:1
 !
 address-family ipv4
 exit-address-family
!
vrf definition data-vrf2
 rd 103:1
 !
 address-family ipv4
 exit-address-family
!
vrf definition voice-vrf1
 rd 101:1
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
```

Create dhcp pool for vrf end-points

```
ip dhcp pool voice-vrf1
 vrf voice-vrf1
 network 8.0.0.0 255.255.0.0
 default-router 8.0.0.1
 option 150 ip 8.0.0.1
!
ip dhcp pool data-vrf1
 vrf data-vrf1
 network 7.0.0.0 255.255.0.0
 default-router 7.0.0.1
 option 150 ip 7.0.0.1
!
!
!
ip name-server vrf voice-vrf1 9.44.44.9
ip name-server 9.44.44.9
```

Create voice VRF and enable voip services

```
voice vrf voice-vrf1
!
voice service voip
 no ip address trusted authenticate
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback
none
 h323
 sip
  session transport tcp
  registrar server
  no silent-discard untrusted
  no call service stop
```

Cisco Unified CME VRF design guide: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/vrf/design/guide/vrfDesignGuide.html

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Cisco CME SIP VRF configuration

Configure CME SIP registrar

```
voice register global
 mode  cme
 source-address 9.42.29.101 port 5060
 max-dn 100
 max-pool 100
 tftp-path flash:
 create profile sync 0067045007438025
 group 1 vrf voice-vrf1
  source-address 8.0.0.1
 !
 group 2 vrf data-vrf1
  source-address 7.0.0.1
 !
 group 3 vrf data-vrf1
  source-address 10.104.45.142
 !
 voice register dn  1
 number 14001
 name voicevrf-ph1
!
voice register dn  2
 number 14002
 name datavrf-ph1
```

```
voice register pool  1
 id mac A40C.C395.7B5C
 session-transport tcp
 type 9971
 number 1 dn 1
 group 1
 template 1
 dtmf-relay rtp-nte
 username 14001 password 14001
 codec g711ulaw
 paging-dn 99
!
voice register pool  2
 logout-profile 1
 id mac ACA0.16FC.9742
 type 9971
 group 2
 dtmf-relay rtp-nte
 codec g711ulaw
 paging-dn 99
```


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Cisco Unified Enhanced SRST

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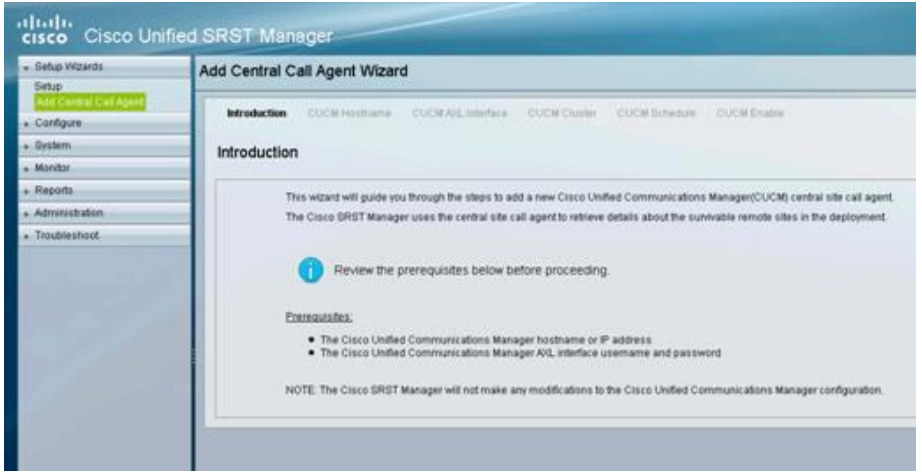
Cisco Unified SRST Manager

- SRST Manager is a Cisco application that runs on a Linux based virtual machine
- Connects to Cisco UCM database and reads configuration and translates configuration to CLI and provisions SRST routers
- Leverages CME as SRST to deliver better user experience during WAN outage
- SRST manager is compatible with Cisco UCM and Cisco Business Edition 6000 versions 8.6, 9.X (upcoming SRST Manager release will support CUCM 10.X)


Cisco Unified SRST Manager
Setup Wizards
Setup
Add Central Call Agent
Configure
System
Monitor
Reports
Administration
Troubleshoot
Add Central Call Agent Wizard
Introduction
Introduction
This wizard will guide you through the steps to add a new Cisco Unified Communications Manager(CUCM) central site call agent.
The Cisco SRST Manager uses the central site call agent to retrieve details about the survivable remote sites in the deployment.
Review the prerequisites below before proceeding.
Prerequisites:
The Cisco Unified Communications Manager hostname or IP address
The Cisco Unified Communications Manager AXL interface username and password
NOTE: The Cisco SRST Manager will not make any modifications to the Cisco Unified Communications Manager configuration.

- Automates CLI provisioning of SRST site routers, greatly impacting OPEX and improving SRST site resiliency
- Supports classic SRST and Enhanced SRST provisioning

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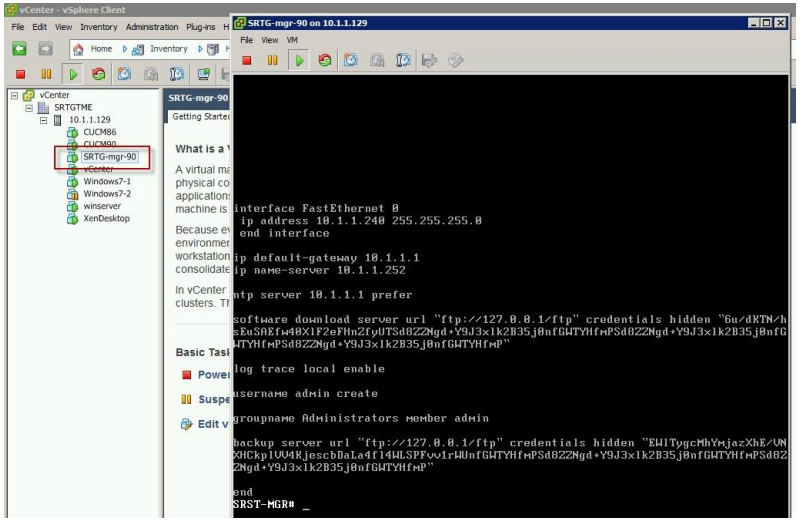
Cisco Unified SRST Manager

Virtual machine

- Supported on VMware ESXi 4.1, 5.0 and 5.1 (5.5 roadmapped)
- SRST Manager virtual machine is delivered in a .ova file. The .ova file includes the SRST manager application software.
- Download: www.cisco.com/go/srst
- Virtual machine reserved resources:
  - 1 vCPU
  - 1GB memory
  - 25GB storage min. (thin provision is recommended to allow for growth as needed, 256GB max.)
  - 1 vNIC
  - OVA file will automatically configure these setting when deployed
- Runs on Cisco UCS, UCS-E and any third-party virtualized server
- Supported on UC on UCS TRC's (Tested reference configurations)



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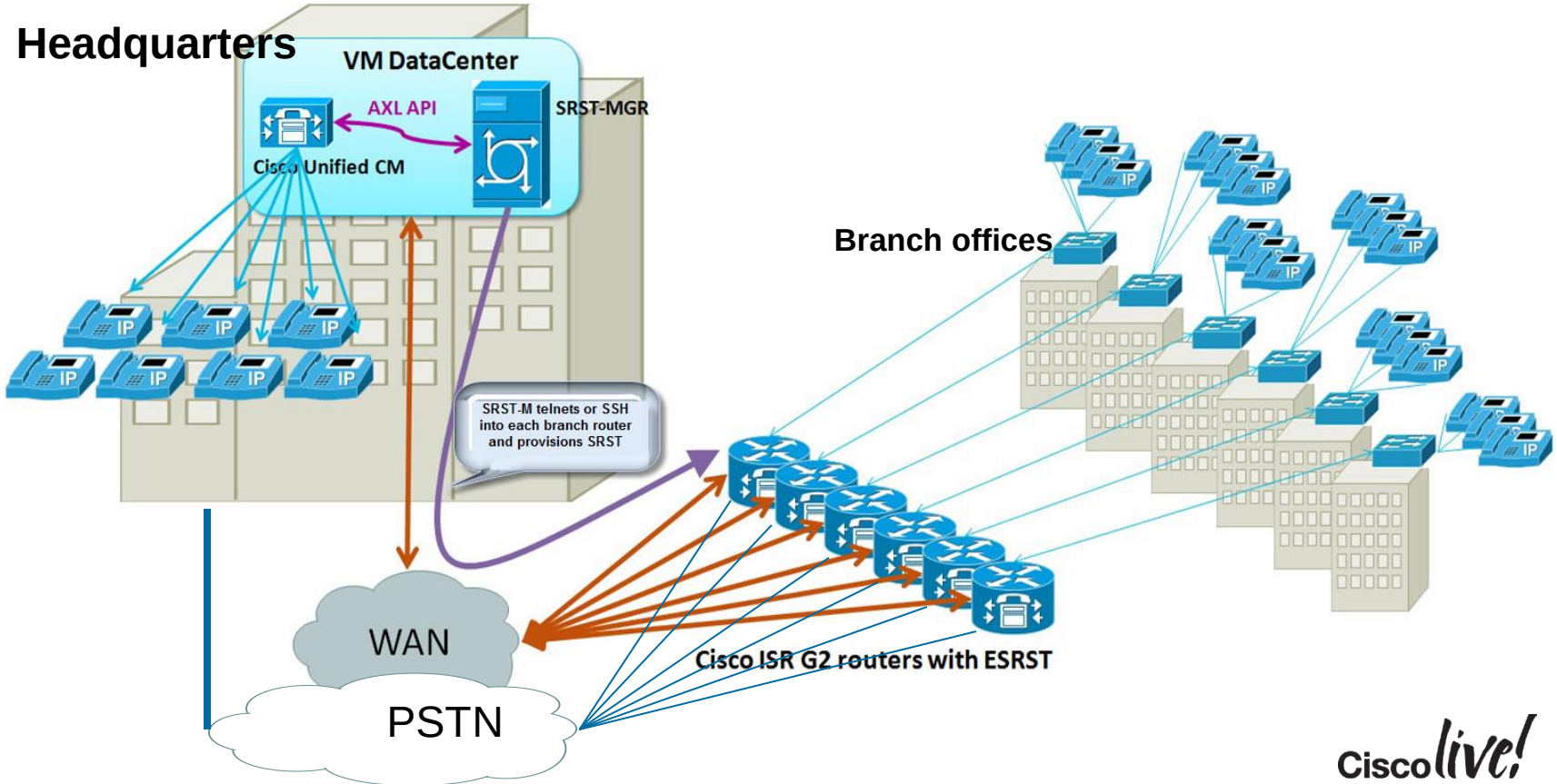
Cisco Unified SRST manager

Configuration flow



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Cisco Unified SRST Manager

Performance

- In subsequent updates SRST Manager looks for any modified, added, erased changes in CUCM. If any, it will push the new configuration; if none, it leaves SRST router alone.
- SRST configuration updates can be scheduled through the provisioning scheduler based on time of day and can be set to perform updates on a daily, weekly or monthly basis
- After SRST manager retrieves branch sites configuration from CUCM it waits for the provision the sites. SRST manager will provision 5 sites at a time (5 simultaneous threads) until all branch sites are configured
- SRST manager bandwidth utilization of a single branch router provisioning is ~100kbps peak (1000 sites, 30 users per site)

Note: Although SRST Manager does not over use network and computing resources it is recommended to run updates during off-peak hours because CUCM queries may cause CUCM processing interruptions. Although CUCM 8.0 and above is designed to ignore CPU request other than call signaling request when CPU utilization crosses a threshold.

SRST Manager can run on 1vCPU or assign 2 vCPU to achieve better performance during first time provisioning and subsequent configuration updates if changes are constant across the majority of sites.

No. of vCPU	500 sites, 30 users provisioned in hours	1000 sites, 30 users provisioned in hours	2000 sites, 30 users provisioned in hours	500 sites, 30 users plus dialplan provisioned in hours	1000 sites, 30 users plus dialplan provisioned in hours	2000 sites, 30 users plus dialplan provisioned in hours
1	5 hrs.	10 hrs.	22 hrs	~6 hrs	~12 hrs.	~26 hrs
2	<3 hrs.	5 hrs.	10 hrs.	<5 hrs	~6 hrs.	~12 hrs.

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Cisco Unified SRST Manager

What's new in Enhanced-SRST

- In Cisco ISR routers there are two existing modes, mode srst or mode cme. Cisco has created a third-mode "mode esrst).
- The "mode esrst" has been developed to support classic SRST scale numbers. This means that a 3945e will be able to support 1500 users with enhanced feature survivability
- The "mode esrst" supports "version negotiation". When SIP phones fall back to SRST mode certain advanced features are disabled on the phones because the features are not supported in SRST mode. With Cisco ESRST mode using version negotiation option IP phone enhanced services are enabled:
  - Shared-Line
  - Busy-Lamp-Field (BLF)
  - Video Calls
- Next SRST Manager release will support REST APIs for management tools to control and provision CUCM and SRST routers. Example: Prime Collaboration or Cisco HCS

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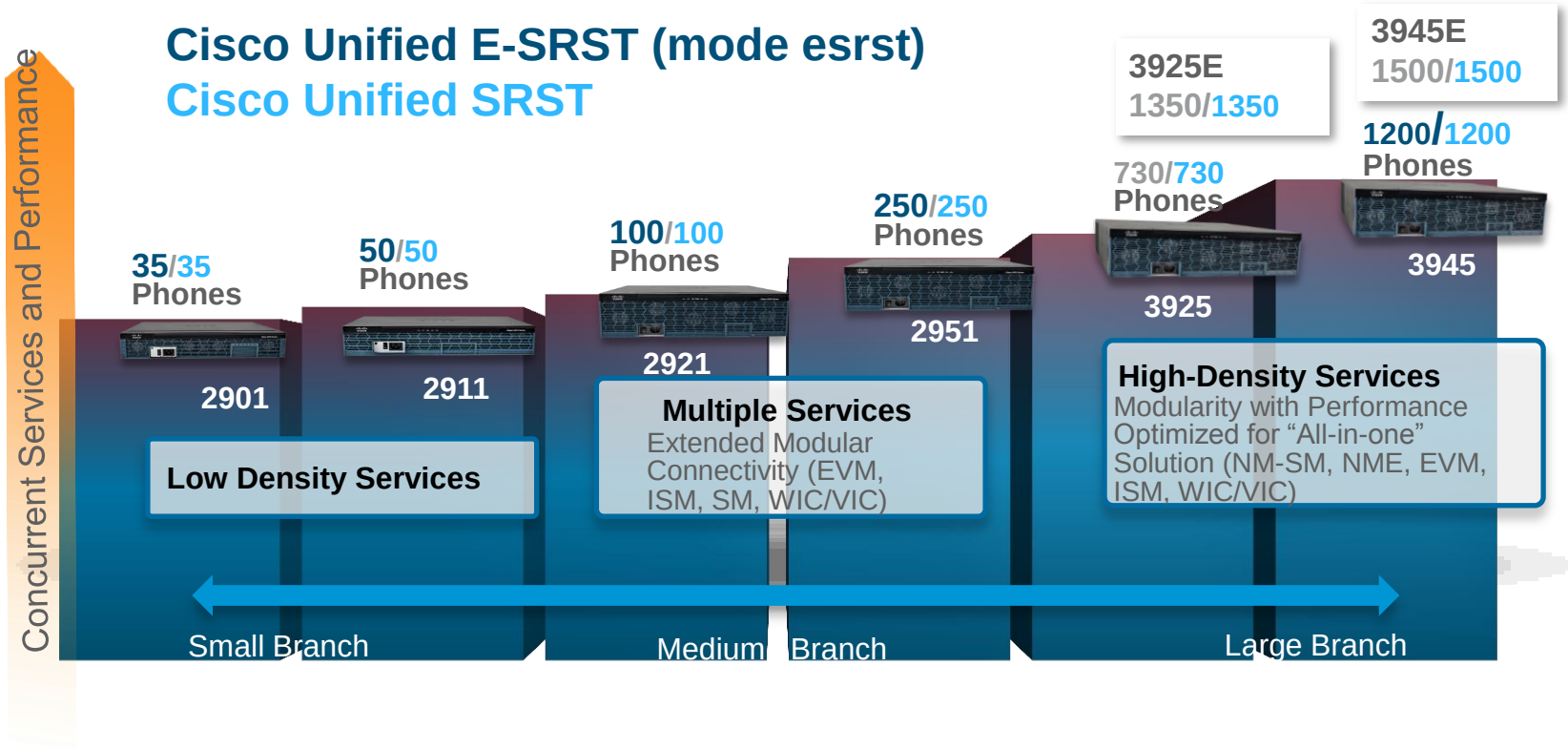
Cisco SRST and E-SRST Capacity on ISR G2



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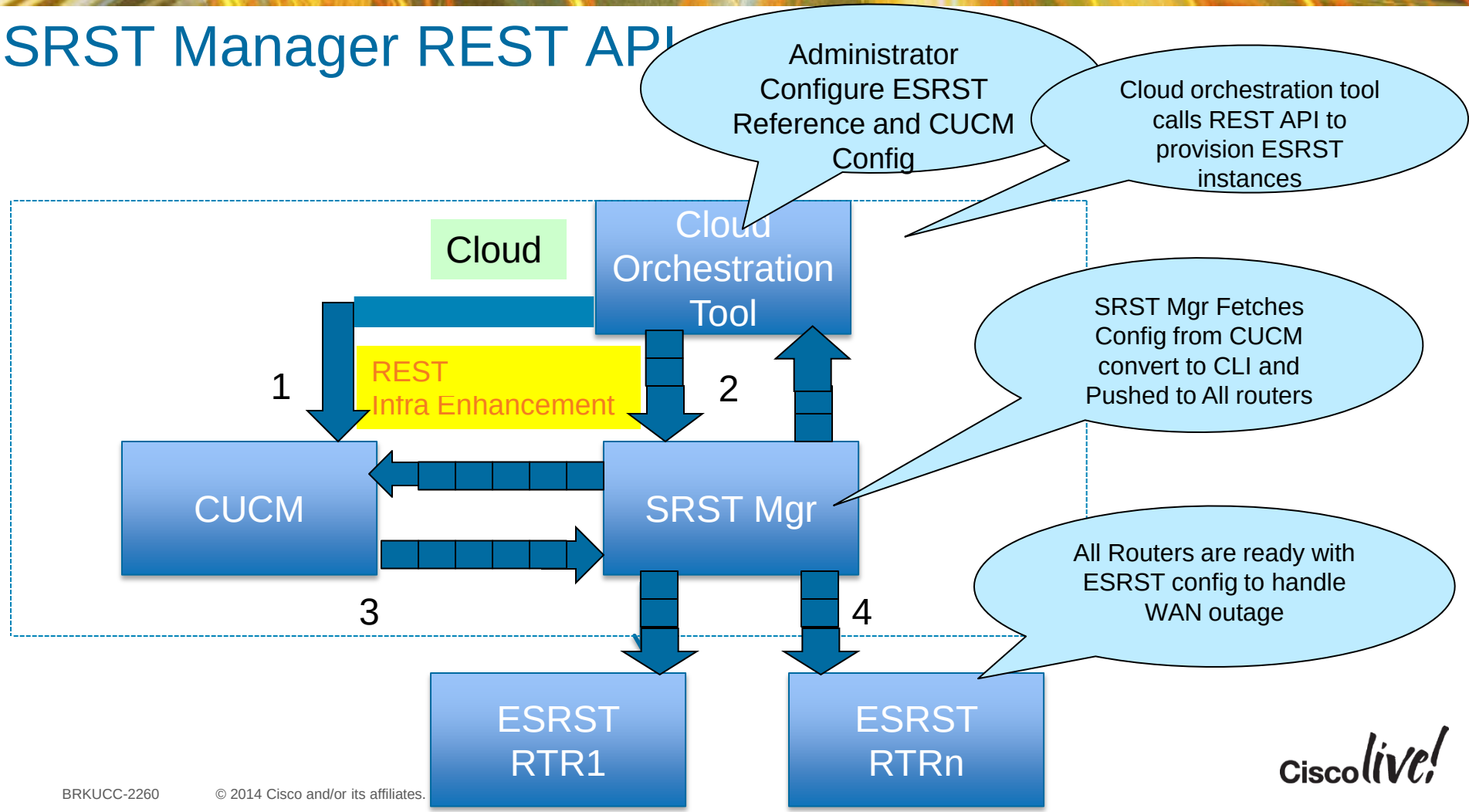
SRST Manager REST API


Administrator Configure ESRST Reference and CUCM Config
Cloud orchestration tool calls REST API to provision ESRST instances
Cloud
Cloud Orchestration Tool
REST Intra Enhancement
1
2
SRST Mgr Fetches Config from CUCM convert to CLI and Pushed to All routers
CUCM
SRST Mgr
3
4
All Routers are ready with ESRST config to handle WAN outage
ESRST RTR1
ESRST RTRn

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SRST Manager DNS requirement workaround

- SRST Manager requires FQDN and DNS integration in order to connect to CUCM AXL database
- This requirement cannot always be met, many customers install CUCM not using AD integration or DNS services on their network
- SRST Manager development team is working on a feature enhancement that will address this caveat, but will not be available for at least six months
- There is a workaround today to allow SRST Manager to connect to CUCM AXL database without Active Directory or DNS services

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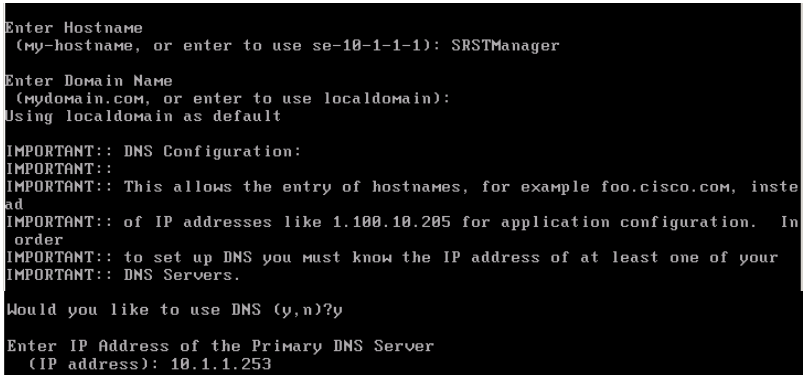
SRST Manager to CUCM AXL connection without AD and DNS workaround

Here are the steps to add CUCM server to SRST manager without AD/DNS services:

1. Install the SRST manager. During the installation you will be asked to set up DNS server, respond 'NO' to that. The default is 'localdomain', so leave it as is.
2. Add the router IP address as the DNS server to SRST Manager


Enter Hostname
(my-hostname, or enter to use se-10-1-1-1): SRSTManager
Enter Domain Name
(mydomain.com, or enter to use localdomain):
Using localdomain as default
IMPORTANT:: DNS Configuration:
IMPORTANT::
IMPORTANT:: This allows the entry of hostnames, for example foo.cisco.com, inste
ad
IMPORTANT:: of IP addresses like 1.100.10.205 for application configuration. In
order
IMPORTANT:: to set up DNS you must know the IP address of at least one of your
IMPORTANT:: DNS Servers.
Would you like to use DNS (y,n)?y
Enter IP Address of the Primary DNS Server
(IP address): 10.1.1.253

3. Download the certificate on the CUCM, you can find it in 'Cisco Unified OS Administration' mode, go to Security ----> Certificate Management tab, you can download the 'tomcat' certificate.
4. Upload this certificate to the SRST Manager.
5. Use a router that SRST manager can reach across the network. Add DNS server config on it as below. For the purpose of this example the hostname of the CUCM is 'CUCM86' and IP address is '172.19.153.155'.

```
Router#conf t
Router(config)#ip host CUCM86 172.19.153.155
Router(config)#ip dns server
        Router(config)#ip domain name localdomain
```

6. Now ping the CUCM from the router using the hostname and the ping should work successfully.
7. Perform the steps to integrate CUCM AXL database connection to SRST Manager

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UC on Cisco UCS E-series

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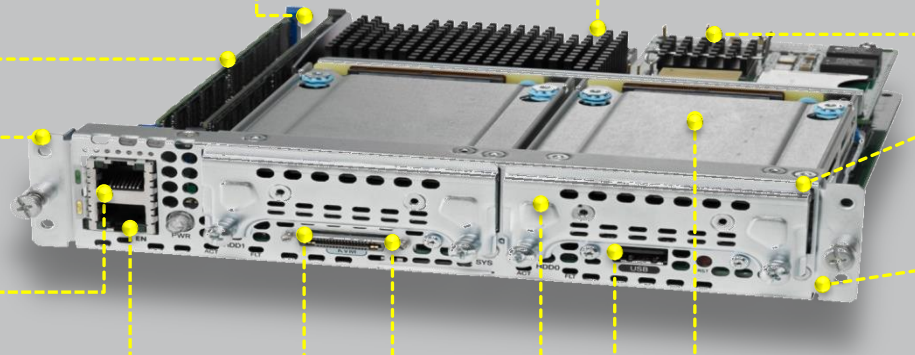
Cisco UCS E-Series Single Wide Blade

Compact, Multipurpose Blade Housed in ISR G2



8, 12, and 16 GB DRAM Options

Maximum 65 W Power Draw 80 Percent Less than Server

Intel Xeon E3 Family Quad-Core Processor

iSCSI Initiator Hardware Offload

Configuration and Management Through CIMC

Remote and Schedulable Power Management

Two SD cards: One for the CIMC and Temporary Storage of OS and One for a Blank Virtual Drive

One External and Two Internal GE Ports

Up to 2 SATA, SAS or SSD Hard Drives

10/100 Ethernet Management Port

USB 2.0 Port for External Device Connectivity

KVM Console Connector

On-Board Hardware RAID 0/1 with Hot-Swap Capability

Wire-Free, Plug-and-Play Modularity, Low Shipping Weight (2.5 lb/1.1 kg)

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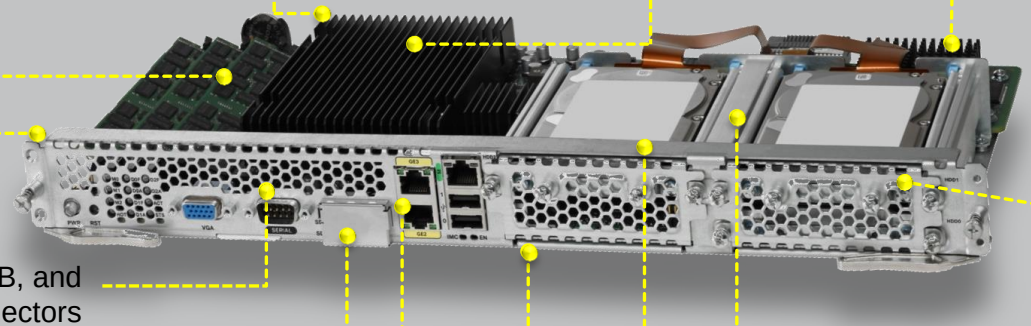
Cisco UCS E-Series Double Wide Blade

Compact, Multipurpose Blade Housed in ISR G2



8 GB - 48 GB DRAM Options

Maximum 130 W Power Draw, 80 Percent Less than Server

Intel Xeon E5-2400 Quad Core or Six-Core Processor

iSCSI Initiator Hardware Offload

Remote and Schedulable Power Management

Out-of-Band Configuration and Management Through CIMC

Front-Panel VGA, 2 USB, and Serial Console Connectors

Up to 3 SATA, SAS, SSD Hard Drives or 2 HDD and a PCIe Card

Two SD Cards: One for the CIMC and Temporary Storage of OS and One for a Blank Virtual Drive

On-Board Hardware RAID 0, 1, and 5 Configuration Options with Hot-Swap Capability

Two External and Two Internal GE Ports with TCP/IP Acceleration

Wire-Free, Plug-and-Play Modularity, Low Shipping Weight (7 lb / 3.2 kg)

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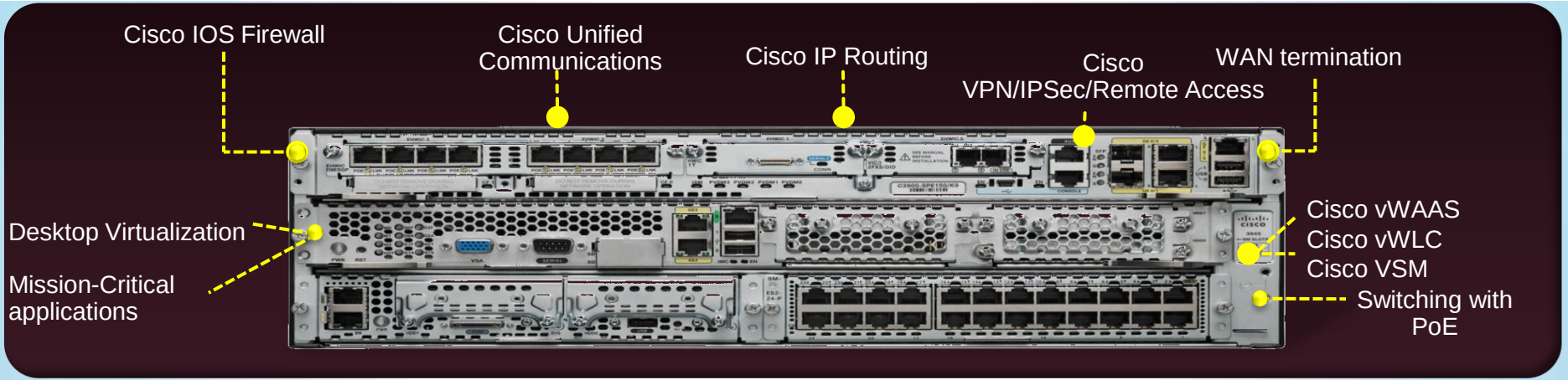
Cisco ISR G2 as Blade Server Enclosure

Use Slots on Most Widely Deployed Branch Device



All-in-One Device for Branch Services

- WAN Optimization
- Wireless LAN/WAN
- Routing/Switching
- Application Hosting
- Unified Communications
- Security

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Cisco Unified Communications applications supported on Cisco UCS E-series

Call Control

- Cisco Unified Communication Manager (spec-based only)
- Cisco VCS Expressway

Voicemail

- Cisco Unity Connection
- Cisco Unity Connection-SRSV

Other UC applications

- Cisco Mediasense network-based recording
- Cisco CUPS – IM/P services

To learn more about Cisco UCS E-series please attend:

PSORST-2005 - Transforming the Branch Office with a Single Box Solution with Kishan Ramaswamy
Wednesday at 11:30am at Moscone West 2001

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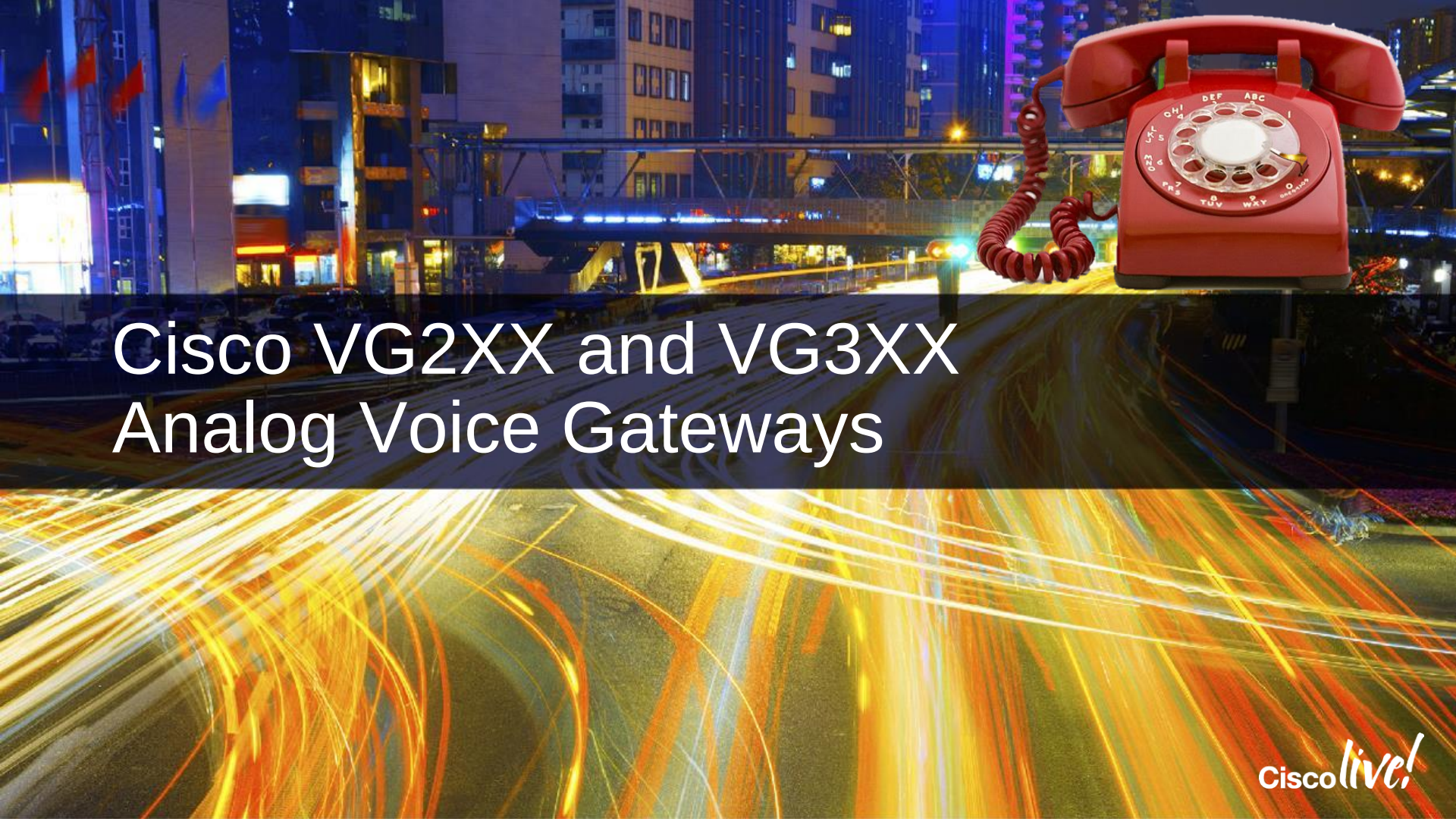

Cisco VG2XX and VG3XX Analog Voice Gateways

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Cisco Analog Voice Gateway Portfolio

Port Density

VG350

High Density Analog Gateway

96 – 160 Ports

NEW

Release planned for Q3CY14

VG310/VG320

24/48 Ports

VG224

VG204XM

VG202XM

Platforms

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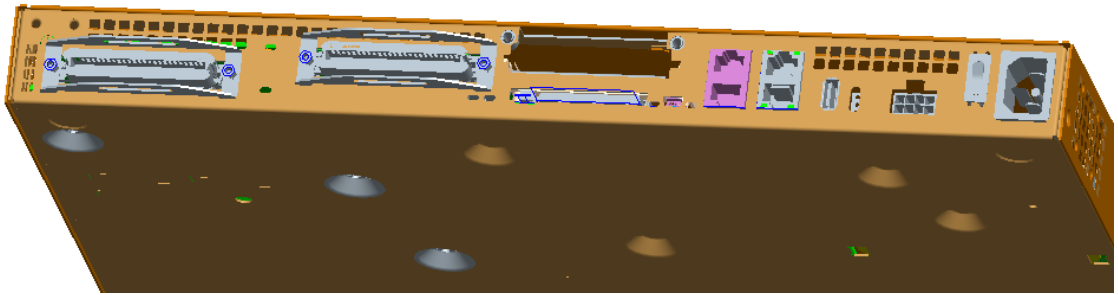
Cisco VG310/VG320

- VG310 and VG320 will be feature parity with VG350
- Fixed integrated FXS ports

August 14' Launch

Feature	Description
VG310 – 24 FXS Port VG	**RoHS 6, 24 port, 1 RU analog voice g/w**
VG320 – 48 FXS Port VG	**RoHS 6, 48 port, 1 RU analog voice g/w**
1 EHWIC slot on VG310/320	**Support T1 CAS,PRI, BRI, FXS, FXO VIC2/VIC3 cards**
OPX-Lite ports on VG310/320	**# of extended loop length ports (8 and 4, respectively)**
Platform	**Based on ISR G2 2901 hardware**



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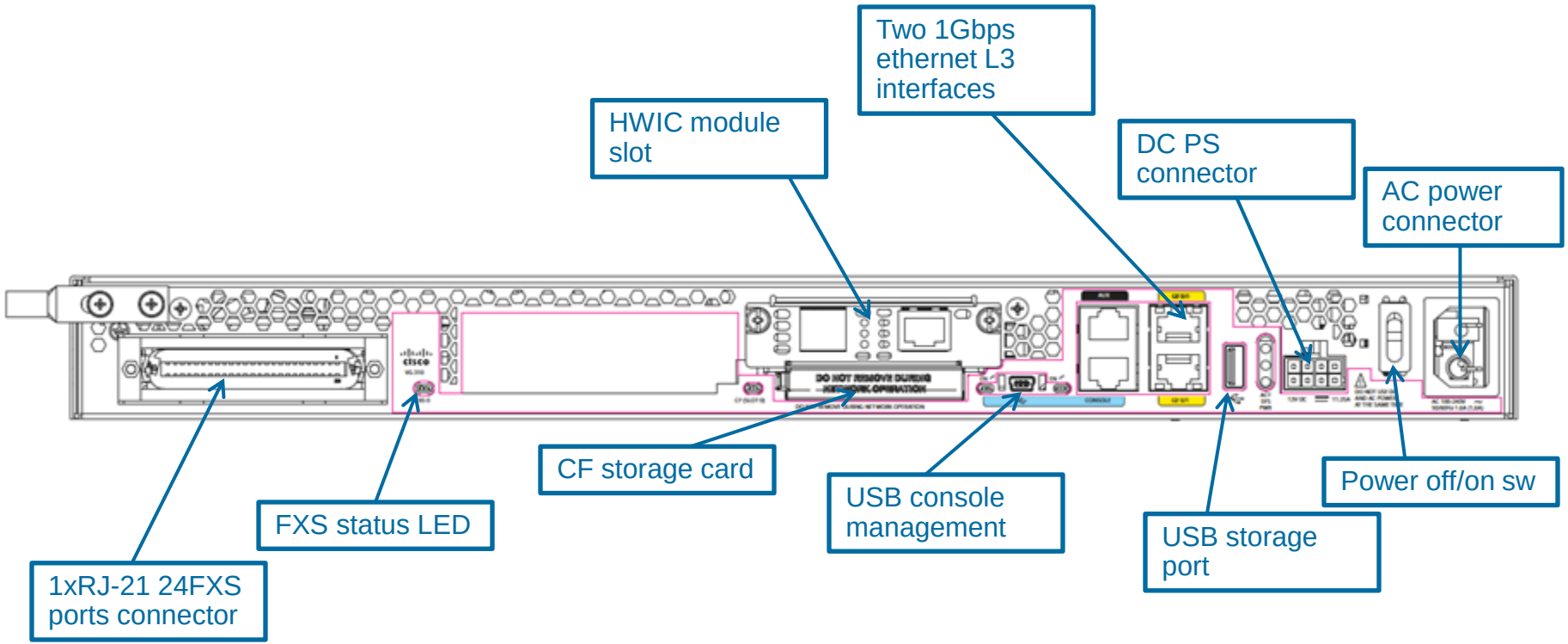
Cisco VG310



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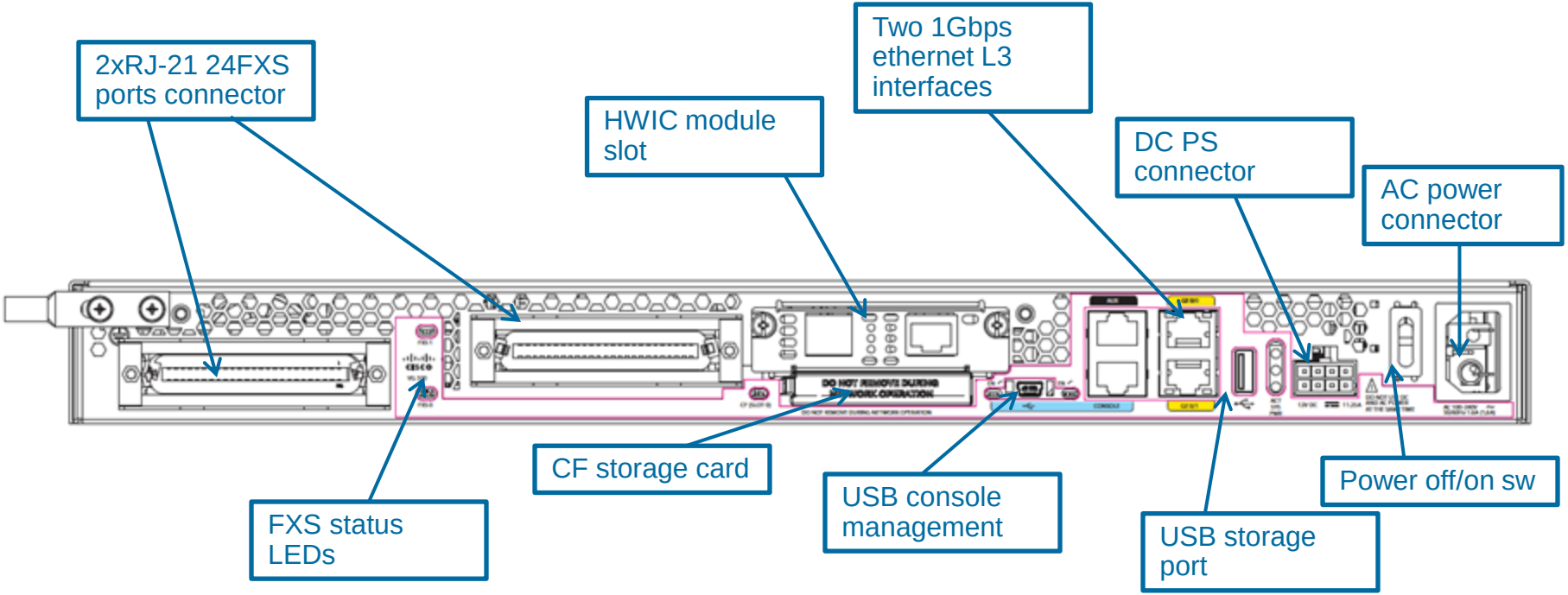
Cisco VG320



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Analog Voice Gateway Benefits & Advantages

- Low price of analog phones
- High cost of replacing cabling
- Harsh environments
- Regulations…
  - Hospitality: 1 analog phone per hotel room
  - Health Care: Phone goes away with the patient
- Do not consume power
- Do not consume switch resource
- Ability to retain a large portion of your analog phone install base while migrating to IP Telephony

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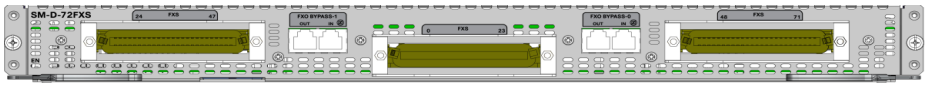
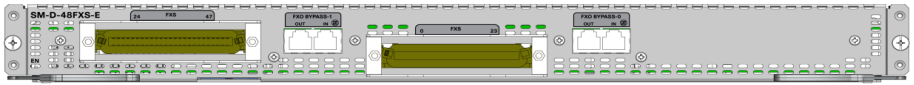
VG350 High Density Analog Gateway Solution



SM-D-72FXS

72 Port FXS Double Wide Service Module



SM-D-48FXS-E

48 Port FXS OPX-Lite Double Wide Service Module

VG350

High Density Analog Gateway

Configuration	VG350		
	SM 1	SM 2	Total num of ports
1	SM-D-72FXS	SM-D-72FXS	144
2	SM-D-72FXS	SM-D-48FXS-E	120
3	SM-D-48FXS-E	SM-D-48FXS-E	96

HWIC slots will support FXS, FXS-E and FXO interfaces

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Cisco High density voice gateway VG350

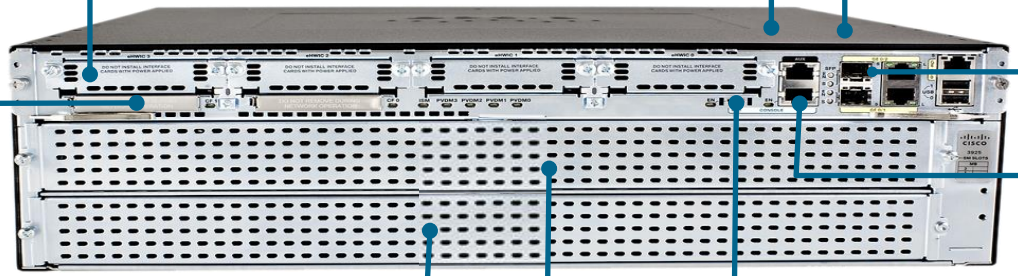
4 EHWIC Slots

- **Support additional** FXS and FXO VICs

PVDM3 DSP Modules

- **Latest and greatest DSP technology**
- Noise Cancelation
- Acoustic Shock Prevention

Redundant Power Supply Option



2 Flash storage slots

3 GE Ports

Standard console and aux ports

2 Double Wide Service Module Slots

- **High Density Analog DWSM Slots**
- Support
  - 72 Port FXS DWSM
  - 48 Port OPX-Lite FXS DWSM

USB Port

- **Console over USB**
- **Convenient Storage**

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Cisco VG350 Highlights

- Modified ISR G2 3945 dedicated for Voice Gateway function
- Software feature parity with VG2xx
  - Voice, fax and modem
  - Comprehensive set of codecs
  - SIP, H.323, SCCP and MGCP protocols
  - TLS / SRTP
  - Supported with CUCM 8.6.2 (SU2), 9.0.1, 9.1 and 10.x
  - Supported with CUCME / SRST 7.1, 8.0, 8.1, 8.5, 8.6, 8.8, 9.0, 9.1, 9.5 and 10.0
- IOS Release: 15.2(4)M or higher
- Supported on Cisco Unified Operations Manager 9.0, see CPUOM release notes for details http://www.cisco.com/en/US/partner/docs/net_mgmt/prime/unified_operations_manager/9.0/Device_Support/OMdevspt90.html

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Cisco VG3XX Highlights

- EHWIC slots can be used for…
  - Additional FXO and FXS ports
    - VIC2-xFXO
    - VIC3-xFXS/DID
    - VIC3-2FXS-E/DID (Long loop length)
  - T1/E1 ports for clock sourcing (in and out)
    - VWIC3-xMFT-T1/E1
- Supports EnergyWise for power savings
- Supports Online Insertion and Removal (OIR) of DWSMs (only VG350)
- Redundant Power Supply option
- Lightening protection
- Cable-detection test verification

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Cisco VG3XX is NOT…

- Not a router
- Does not support T1/E1 voice. The T1/E1 ports can be used only for clock sourcing (in and out)
- Does not support SRST natively
- Does not support CME natively
- Does not support any WAN interface modules
- Does not support WLAN modules

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VG3XX at-glance features

Applications	IOS Based Unified Communication Analog Gateway solution for UC high-density analog deployments
Telephony Interfaces	FXS and FXO
Voice Codecs	G.711, G.722, G.726, G.728, G.729, iLBC, iSAC
Call Control Protocols	SCCP, MGCP, SIP, H323
Call Agents	Cisco UCM, Cisco UCME
Fax for SIP and H323	standard-based T.38 fax relay, NSE based T.38 fax relay, fax pass-through and modem relay
Fax for MGCP	standard-based T.38 fax relay, NSE based T.38 fax relay and modem relay
Fax for SCCP	Modem-passthrough (works for fax pass-through applications), Modem Relay and NSE based T.38 fax relay
Encryption	SRTP/TLS, IPsec
Data Interfaces	3 Gigabitethernet interfaces
Data Features (For redundant link configuration ONLY)	EIGRP, OSPF, RIP, NAT, IRB

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VG3XX Service Module and HWIC modules

Available slots	Supported modules
SM slot 2 (VG350 only	SM-D-72FXS, SM-D-48FXS-E
SM slot 4 (VG350 only)	SM-D-72FXS, SM-D-48FXS-E
EHWIC 0	VIC2-2FXO, VIC2-4FXO, VIC3-2FXS/DID, VIC3-2FXS-E/DID, VIC3-4FXS-E/DID
EHWIC 1	VIC2-2FXO, VIC2-4FXO, VIC3-2FXS/DID, VIC3-2FXS-E/DID, VIC3-4FXS-E/DID
EHWIC 2	VIC2-2FXO, VIC2-4FXO, VIC3-2FXS/DID, VIC3-2FXS-E/DID, VIC3-4FXS-E/DID
EHWIC 3	VIC2-2FXO, VIC2-4FXO, VIC3-2FXS/DID, VIC3-2FXS-E/DID, VIC3-4FXS-E/DID

VG350 motherboard comes pre-installed with one PVDM3-256, remaining PVDM slots can be populated with PVDM3 modules only

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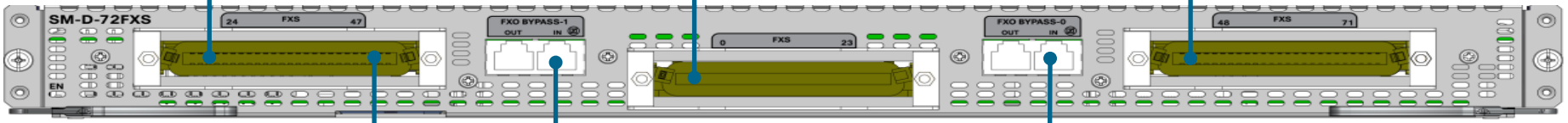
72 Port FXS DWSM
SM-D-72FXS

Ports 24 – 47
- **Ports 46 & 47 connected to PSTN FXO Bypass failover ports**

Ports 0 – 23
- **Ports 0 – 3 support long loop length**

Ports 48 – 71



Standard RJ21 Connectors

- SM-D-72-FXS
  - 72 port FXS Double Wide Service Module
  - Loop length – 3,300 ft with 26 AWG
  - First 4 ports will support OPX-Lite long loop length
    - 11,000 ft with 26 AWG
    - 18,000 ft with 24 AWG

2 PSTN FXO Bypass Failover Ports
- Port 46 ←→ **FXO Bypass Port 0**
- Port 47 ←→ FXO Bypass Port 1
- **OUT ←→ FXO Port**
- IN ←→ PSTN

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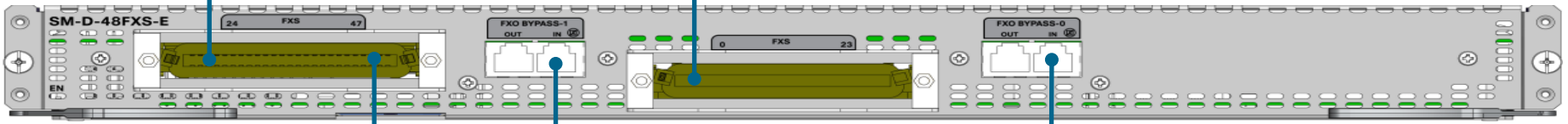
48 Port OPX-Lite FXS DWSM
SM-D-48FXS-E

Ports 24 – 47

- **Ports 46 & 47 connected to PSTN FXO Bypass failover ports**

Ports 0 – 23



2 PSTN FXO Bypass Failover Ports

- Port 46 ←→ **FXO Bypass Port 0**
- Port 47 ←→ FXO Bypass Port 1
- **OUT ←→ FXO Port**
- IN ←→ PSTN

Standard RJ21 Connectors

- SM-D-48-FXS-E
  - 48 port OPX-Lite FXS Double Wide Service Module
  - All ports will support long loop length
    - 11,000ft with 26 AWG
    - 18,000ft with 24 AWG

Long loop-length configuration example:

```
Router#conf t
Router(config)#voice-port 2/0/0
Router(config-voiceport)#loop-length long
Router(config-voiceport)#end
```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Cisco VG3XX long loop-length

- The VG350 FXS-E ports are capable of driving 1400ohms of loop resistance including the phone impedance, subtract 300ohms for phone impedance and you have a maximum of 1100ohms of loop impedance.
- Datasheet loop-length distance specs are based on commonly used 26AWG and 24AWG wire
- Longer distances can be achieved using higher grade wire based on resistance per 1000ft, for example:
  - 22AWG wire has a resistance of 16.14ohms per 1000ft
  - Multiply 16.14 by 2 for a total of 32.3ohms for the tip/ring loop when the circuit is closed
  - 22AWG wire can achieve ~31,000ft of loop-length

Important Note: These calculations are based on the supported DC loop resistance. Other factors related to the quality of the wire lines such as signal attenuation and noise could also impact the maximum loop length capabilities

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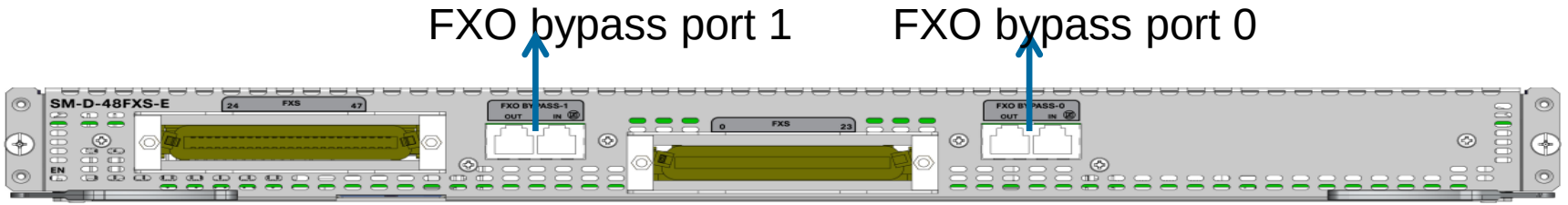
FXO bypass failover



- No configuration required
- Each SM-D-FXS service module support two FXO bypass pairs. FXO bypass pair 0 falls back to FXS port 46 and FXO bypass pair 1 falls back to FXS port 47.
- PSTN FXS line connects to FXO bypass “in” and FXO bypass “out” port connects to an on-premise FXO port termination.
- During normal operation the FXO bypass acts as a wire between PSTN and on-prem FXO port. During power outage or energy savings (scheduled power off) the FXO bypass pairs switch their respective relays to ports 46 and 47 and provide dial-tone from the PSTN
- The FXO on-premise port termination can be a VIC2-FXO card installed on the VG350 HWIC slot, or it can be a VIC2-FXO card installed on a separate gateway.

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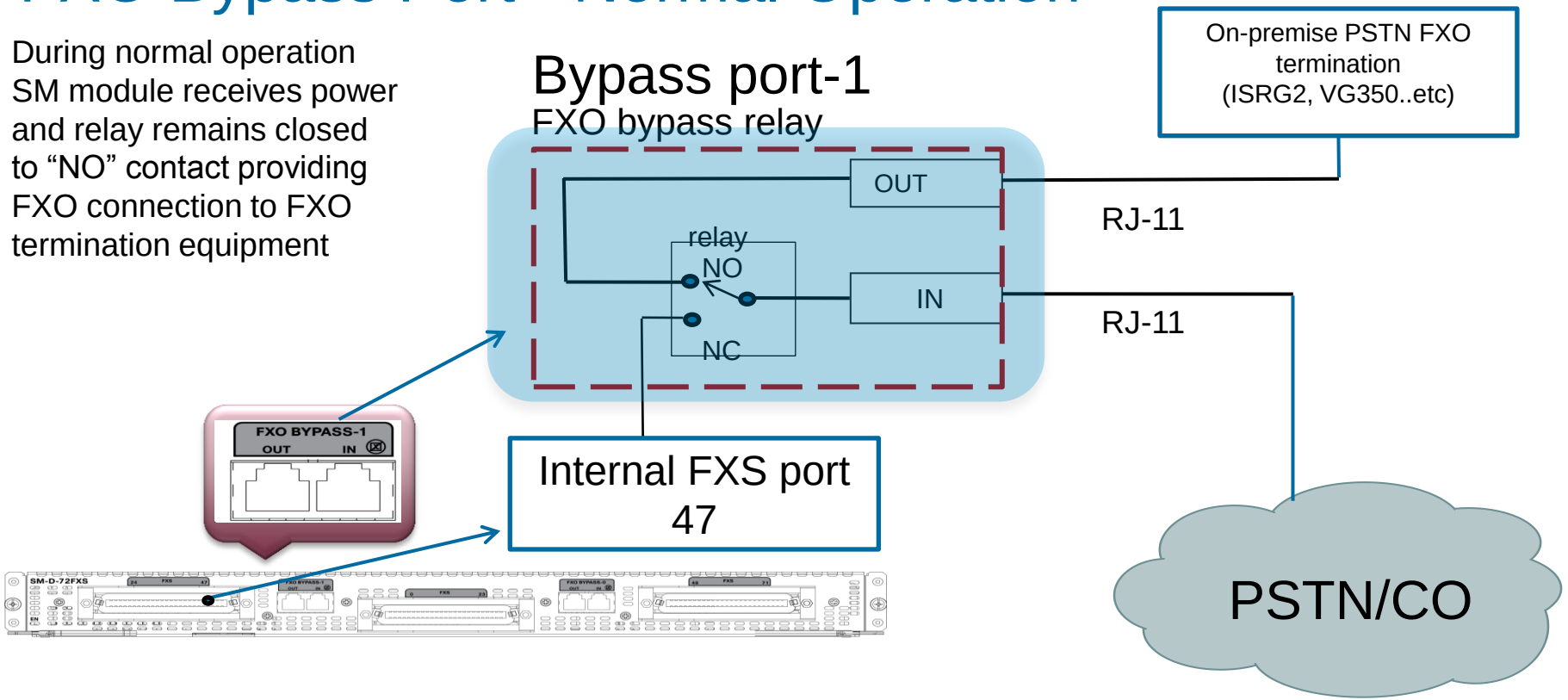
FXO Bypass Port - Normal Operation

During normal operation SM module receives power and relay remains closed to “NO” contact providing FXO connection to FXO termination equipment



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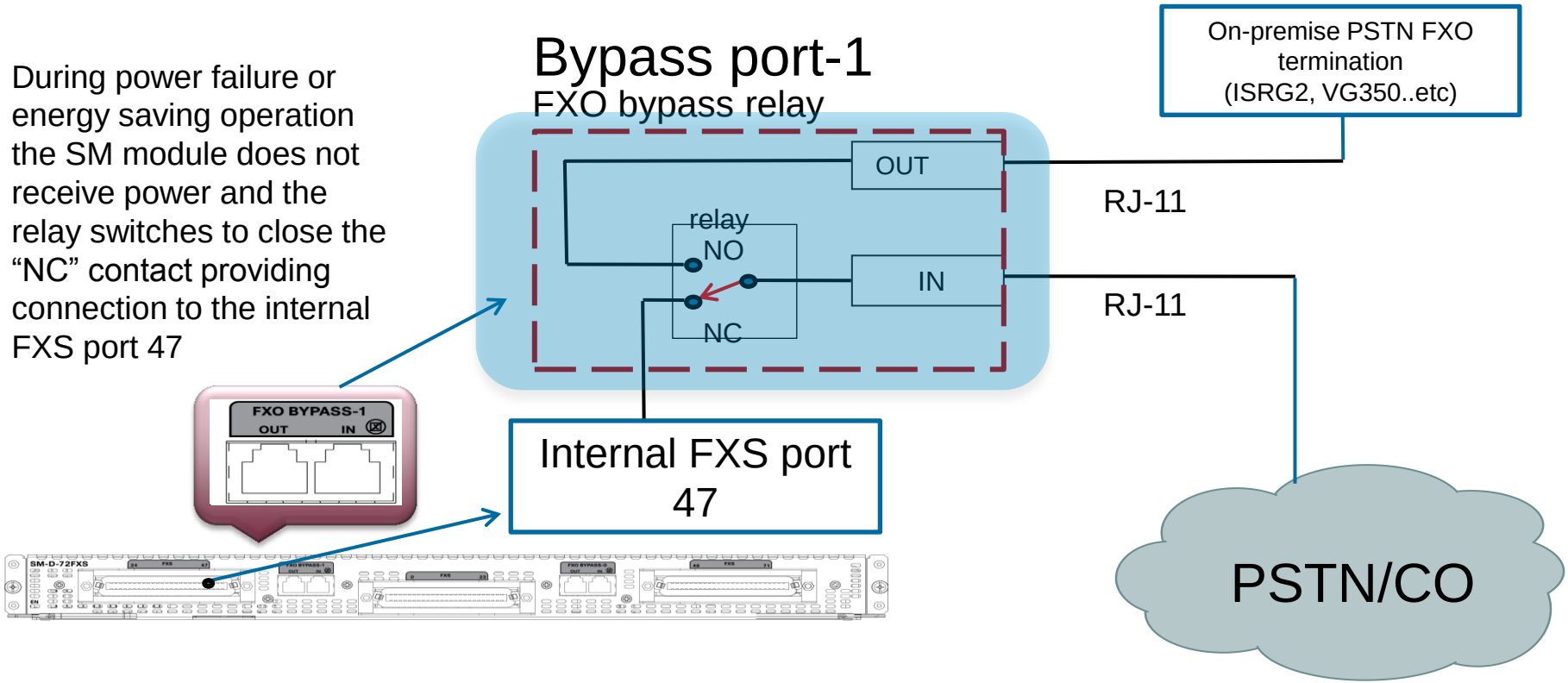
FXO Bypass power fail or energywise operation

During power failure or energy saving operation the SM module does not receive power and the relay switches to close the “NC” contact providing connection to the internal FXS port 47



Note: FXO Bypass-0 port provides fallback connectivity to FXS port 46

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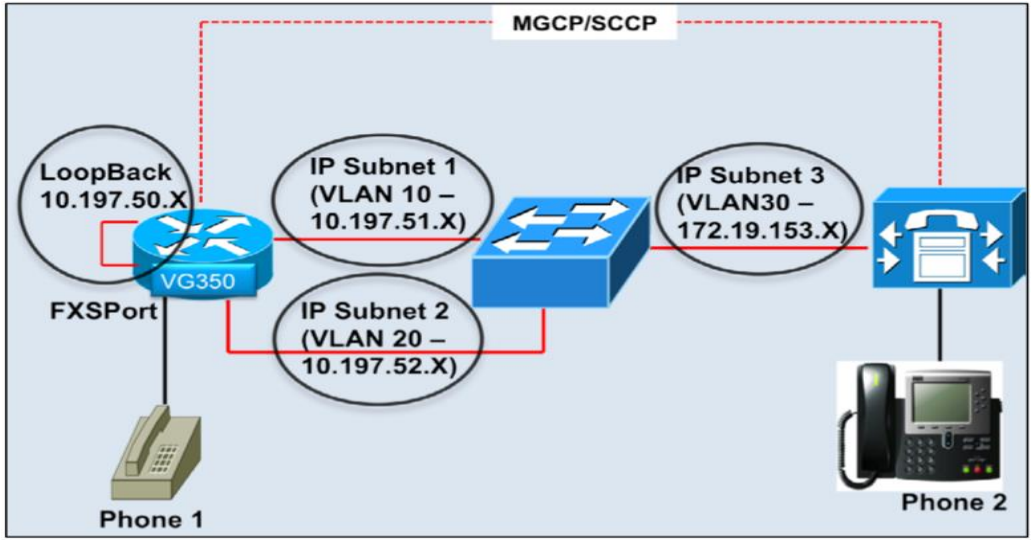
Cisco voice gateway dual-homing



- All VG platforms support two or more ethernet ports
- Create redundant network paths between VG appliance and the call control agent (CUCM, CME)
- Redundant paths are supported using L3 dynamic routing or L2 BVI interface.
- Both SCCP and MGCP are supported for dual-homing

Detailed application note available here:
http://www.cisco.com/c/dam/en/us/td/docs/routers/access/vg224/software/configuration/guide/Cisco_Analog_Gateways_Network_Survivability_Deployment_Options.pdf

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VG3XX cable detection

- The VG350 has the ability to run a cable detection test and can be invoked via a background task with a configured polling timer or CLI to manually invoke the cable detect test.
- The cable detection test is supported on each voice-port
- The CLI to configure cable detect polling timer for background task :

Global config:

VG350(config)#voice service pots

VG350(config-voi-serv)# [no] cable-detect-poll-timer <0-1440> in minutes

- CLI to manually invoke cable detect test:

VG350#show voice cable <X/Y/Z> “<voice port to be tested>”

To avoid conflict, this CLI requires cable-detect polling timer disabled first.

- The administrator can capture the test events to SNMP traps

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VG350 Power Consumption

Chassis configuration: VG350 with One SM-D-72FXS and One SM-D-48FXS-E					
All ports idle (no VIC cards inserted)	10 active calls/5 ports ringing (no VIC cards inserted) 12% capacity	20 active calls/10 ports ringing (no VIC cards inserted) 25% capacity	40 active calls/ 20 ports ringing (no VIC cards inserted) 50% capacity	80 active calls/ 40 ports ringing (no VIC cards inserted) 100% capacity (more active calls)	40 active calls/ 80 ports ringing (no VIC cards inserted) 100% capacity (less active calls)
150 watts	155 watts	159 watts	166 watts	181 watts	173 watts

Chassis configuration: VG350 with Two SM-D-72FXS, short loop-length			
All ports idle (no VIC cards inserted)	40 active calls/40 ringing (no VIC cards inserted)	72 active calls/72 ringing (no VIC cards inserted)	160 active calls (includes 4-VIC3-4FXS)
163 watts	190 watts	220 watts	278 watts

Chassis configuration: VG350 with Two SM-D-48FXS-E, long loop-length			
All ports idle (no VIC cards inserted)	24 active calls/24 ringing (no VIC cards inserted)	48 active calls/48 ringing (no VIC cards inserted)	112 active calls (includes 4-VIC3-4FXS)
140 watts	213 watts	268 watts	379 watts

VIC-FXS cards add ~2 watts
VIC-FXS-E cards add ~4 watts in active state

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VG350 and Energywise

- VG350 supports Energywise power saving features
- Power savings delivered through recurring scheduled shutdown of service modules
- During service module shutdown FXO bypass ports provide dial-tone survivability
- Configuration CLI is consistent with ISR G2
- Supported modules are SM-D-72FXS, SM-D-48FXS-E and PVDM3

Energywise configuration example:

```
Router(config)#energywise domain <variable tag> security shared-secret 0 <variable tag>

Router(config)#hw-module sm 2
Router(config-hw)#energywise level 0 ?

 recurrence  Specify that this level setting should recur at the specified time(s)

Router(config-hw)#energywise level 0 ➔ '0' is defined as "shutdown"
```

For more information visit: http://www.cisco.com/en/US/docs/routers/access/1900/software/configuration/guide/enrgyz_artg.html

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VG350 Energywise savings

One SM-D-72FXS and one SM-D-48FXS-E

All SM and PVDM powered up	Module slot 2 (SM-D-72FXS) shutdown	Module slot 2 (SM-D-72FXS) and 4 (SM-D-48FXS-E) shutdown	Module slot 2(SM-D-72FXS), 4 (SM-D-48FXS-E) and PVDM shutdown
150 watts	110 watts	83 watts	79 watts

Two SM-D-72FXS

All SM and PVDM powered up	Module slot 2 (SM-D-72FXS) shutdown	Module slot 2 (SM-D-72FXS) and 4 (SM-D-48FXS-E) shutdown	Module slot 2(SM-D-72FXS), 4 (SM-D-48FXS-E) and PVDM shutdown
163 watts	**123 watts**	**84 watts**	**80 watts**

Two SM-D-48FXS-E

All SM and PVDM powered up	Module slot 2 (SM-D-72FXS) shutdown	Module slot 2 (SM-D-72FXS) and 4 (SM-D-48FXS-E) shutdown	Module slot 2(SM-D-72FXS), 4 (SM-D-48FXS-E) and PVDM shutdown
140 watts	**111 watts**	**84 watts**	**80 watts**

VIC voice cards add ~2 watts per HWIC slot in idle state
VIC voice cards are not supported by energywise

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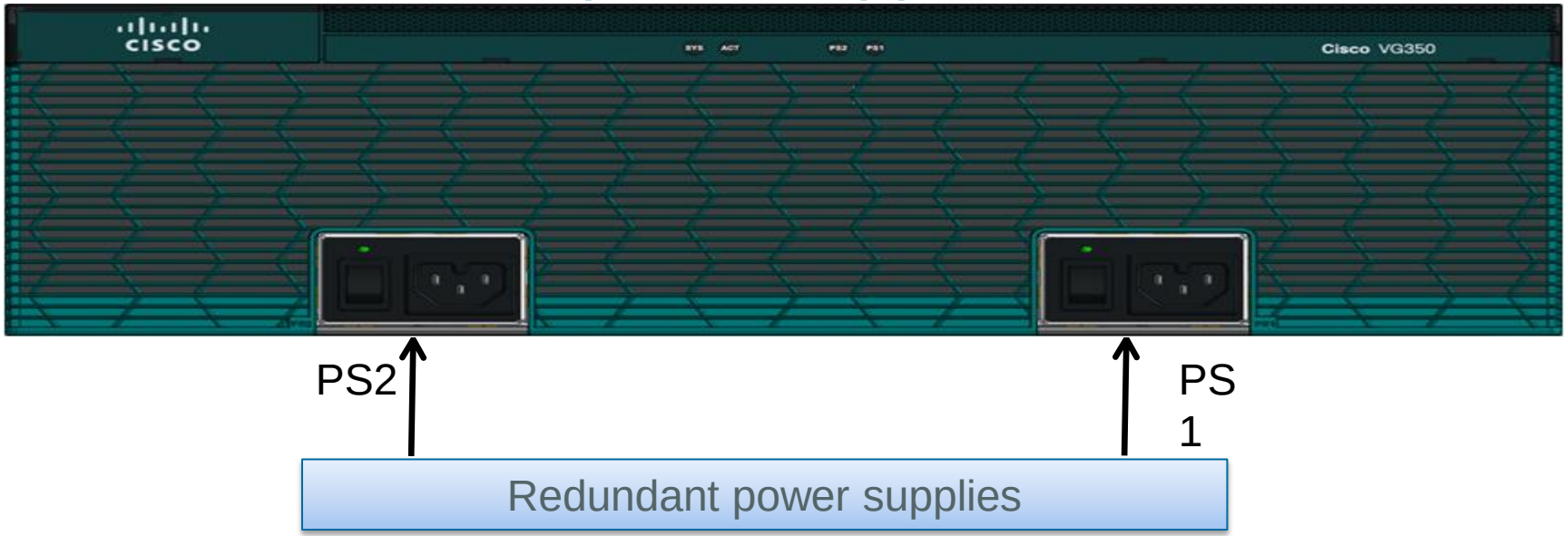
VG350 Redundant power supplies for HA



- The supported power supply combinations:
  - AC/AC
  - DC/DC
  - AC/DC power supplies

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Voice Product Feature Comparison

Product	SM-D-72FXS	SM-D-48FXS-E	VIC3-4FXS/DID	VIC3-2FXS-E/DID	VG224
Total DC Loop Resistance (including terminal and phones)	up to 600 ohm	**up to 1400 ohm**	up to 600 ohm	**up to 1400 ohm**	up to 600 ohm
On-Hook Voltage	-44V	-44V	-44V	-44V	-43V
I off-hook max	25mA	**35mA**	25mA	**35mA**	25mA
Vringing	**54 Vrms** to 5REN Zero loop length	**62 Vrms** to 2REN. Zero loop length	45 Vrms to 5REN Zero loop length	**62 Vrms** to 2REN Zero loop length	50 Vrms to 5REN Zero loop length
REN loading per Port	5REN	2REN	5REN	2REN	5REN
Loop Length (26AWG) Loop Length (24AWG)	**3000ft** **5500ft**	**11,000 ft** **18,000 ft**	1500ft	**11,000 ft** **18,000 ft**	3000ft
VMWI support	FSK, **DC Voltage (~100V)**	FSK, **DC Voltage (~100V)**	FSK	FSK	FSK, **DC Voltage** (~83V)

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New CLI for voice-port

voice-port 2/0/0-71
Specifies range of ports

IOS Today

- 6 CLI per port to configure basic features
- 432 commands needed to configure 72 ports

```
Today#config t
Today(config)#voice-port 2/0/0
Today(config-voiceport)#caller-id block
Today(config-voiceport)#ren 2
Today(config-voiceport)#mwi
Today(config-voiceport)# loop-length long
Today(config-voiceport)#station-id number 70020
```

432 CLI commands to configure 72 ports

New IOS CLI

- Voice-port range provisioning
- 6 CLI to configure 72 ports

```
VG#config t
VG(config)#voice-port 2/0/0-71
VG(config-voiceport)#caller-id block
VG(config-voiceport)#ren 2
VG(config-voiceport)#mwi
VG(config-voiceport)#loop-length long
VG(config-voiceport)#station-id number base 20 interval 1 70000
```

6 IOS commands for 72 ports

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New CLI for voice-port -How it works-

- New CLI allows the admin to specify a voice-port range to configure voice-port settings across all ports specified.

  Example:

  VG350#voice-port 2/0/0-20 → New cli enhancement allows for a port range to be defined

- When using the voice-port range configuration the “show running-config” will show all the voice-port configurations as it would show when configuring the voice-ports one-by-one.

- The following commands will be supported for voice-port range mode:

```
VG350(config)#voice-port 0/0/0-71
             VG350(config-voiceport)#?
             battery-reversal    Enable FXS battery-reversal generation
             busyout             Configure busyout trigger event & procedure
             caller-id           Configure port caller id parameters
             default             Set a command to its defaults
             description         Description of what this port is connected to
             disconnect-ack      FXS sending disconnect acknowledge
             exit                Exit from voice-port configuration mode
             mwi                 Enable MWI on this port
             no                  Negate a command or set its defaults
             ren                 Ringer Equivalence Number
             ring                Ring frequency Parameters
             shutdown            Take voice-port offline
             signal              Configure signal parameters for FXS/DID VIC
             snmp                Modify SNMP voice port parameters
             station-id          Configure station ID
```


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New CLI for Dial-peer “pots dial-peer group”

IOS Today

-6 cli per dial-peer to configure basic features
- 432 cli to provision all 72 dial-peers for SM-D-72FXS

```
Today(config)#dial-peer voice 1 pots
Today(config-dial-peer)#description **GROUP1**
Today(config-dial-peer)#port 2/0/0
Today(config-dial-peer)#destination-pattern 1000
Today(config-dial-peer)#preference 0
Today(config-dial-peer)#service mgcp
```

-Same cmd sequence repeated 72 times (432 commands) –

New dial-peer group CLI

- 4 CLI to provision all dial-peers for a SM-D-72FXS
- Each dial-peer is created with a unique destination-pattern

```
VG350(config)#dial-peer group 1
VG350(config-dp-auto)#port 2/0/0-71 destination-pattern 1000 1
description **GROUP 1**
VG350(config-dp-auto)#preference 0
VG350(config-dp-auto)#service mgcp
```


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New CLI for dial-peer -How it works-

- When configuring “dial-peer group” the cli appears in a condensed form in show running-config

```
dial-peer group 10 pots
 port 2/0/0-71 dest 100 1
```

- To show status or detail information use, “show dial-peer group <tag> summary” or “show dial-peer group <tag> detail commands

```
VG350#sh dial-peer group summary
dial-peer group 10 pots:
          AD                          PRE PASS            OUT
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN     FER THRU SESS-TARGET    STAT PORT    KEEPALIVE
21474- pots  up   up          100            0               up   2/0/0
83647
21474- pots  up   up          101            0               up   2/0/1
```

- An administrator can assign STCAPP service to all ports using a single cli command:

```
VG350(config)#dial-peer group 10 pots all stcapp
```

- The Commands available under dial-peer group

```
VG350(config-dp-group)#?
Dial Peer Group commands:
  default      Set a command to its defaults
  description  Dialpeer specific description
  exit         Exit from dial-peer auto-provision configuration mode
  no           Negate a command or set its defaults
  port         Voice port number or range
  preference   Configure the preference order of this dialpeer
  service      The selected service
  shutdown     Change the Admin State to down (no->up)
```


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VG350 SCCP Features

- Feature parity with VG224
- Can register SCCP analog ports to CUCM or CUCME (VG3XX does not support CME natively, there must be a separate CME enabled router on site)
- Supports fallback survivability of sccp analog ports to an SRST router (VG350 does not support SRST natively, there must be a separate SRST enabled router at the branch site)
- Auto-config for SCCP ports with CUCM
- VG350 supports mixed SCCP and MGCP ports on a single gateway
- SCCP supports analog supplementary-services

- Audible Message-Waiting Indication (AMWI)
- Call Forward Busy (CFB)
- Call Forward No Answer (CFNA)
- Call Transfer
- Call Waiting
- Caller ID
- Conference Call
- Distinctive Ringing on Cisco IOS SCCP Voice Gateways
- Feature Access Codes (FACs)
- Redial

- Speed Dial
- Speed Dial to Voice Mail
- Call Forward All (CFA)
- Call Forward All Cancel
- Call Park Directed
- Call Pickup Directed
- Call Pickup of Ringing Extension, Directed
- Call Pickup Group
- Call Pickup Local
- Cancel Call Waiting
- Forward to Voice Mail
- SCCP Meet-Me Conference

http://www.cisco.com/en/US/partner/docs/ios/voice/fxs/configuration/guide/15_0/fxs_15_0_cg.html

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SCCP auto-config

- VG350 supports auto-configuration when provisioned with the Cisco Unified Communications Manager
- The following needs to be completed before powering on the VGs and connecting the device to the network
  - CUCM needs to be provisioned with the VG3XX SCCP port configuration (BATS tool supported)
  - DHCP server will provide the required TFTP Server IP Addresses in the DHCP Offer Option 150 (DNS server and domain name if required)
- VG3XX does a DHCP request and obtains CUCM's IP Address (TFTP address) then contacts the server for the configuration file
- VG self-provisions using the config.xml file obtained from the tftp server and is operational.
- If the network has separate voice and data vlan, the VG should be connected to the voice vlan
- If the deployment does not have a DHCP server an admin can do CLI configuration as explained in the next slide

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SCCP auto-config continued

If DHCP server is available on CCM

```
interface gig0/0
   ip address dhcp
sccp local GigabitEthernet0/0
ccm-manager sccp local GigabitEthernet0/0
ccm-manager sccp
voice service voip
fax protocol t38 nse force
```

If DHCP server is not available

```
interface GigabitEthernet 0/0
  ip address <static ip and mask>
 sccp local GigabitEhernet0/0
 voice service voip
 fax protocol t38 nse force
 ccm-manager sccp local GigabitEthernet0/0
 ccm-manager config server <TFTP IP address>
 ccm-manager sccp
(Following commands are required if CUCM is using FQDN)
ip domain-name
ip name-server
```


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VG3XX Software Packaging

- Single “universalk9” image
  - vg350-universalk9-mz.SPA
- Base software package supports ipbasek9 and uck9 feature sets
  - SSH is included with ipbasek9
- Security feature license needs to be purchased to enable securityk9 feature set
  - SRTP, TLS, IPSEC
- Data feature license needs to be purchased to enable datak9 feature set
  - RIP, EIGRP, OSPF (L3 routing protocols are for L3 link redundancy implementations only)
- vg350-universalk9_npe-mz image is added to support None Payload Encryption (NPE) which does not have VPN and encryption subsystems for countries, such as China and Russia.

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ISR G2 supports SM-D-FXS service modules

- SM-D-72FXS and SM-D-48FXS-E service modules are supported on Cisco 2951, 3925, 3925E, 3945, and c3945E platforms
- On ISR platforms c3945 and c3945E, only one SM-D-FXS double wide service module is supported. (in slots 3 and 4)
- ISR G2 platforms will support all features supported by VG350, plus all ISR G2 feature services

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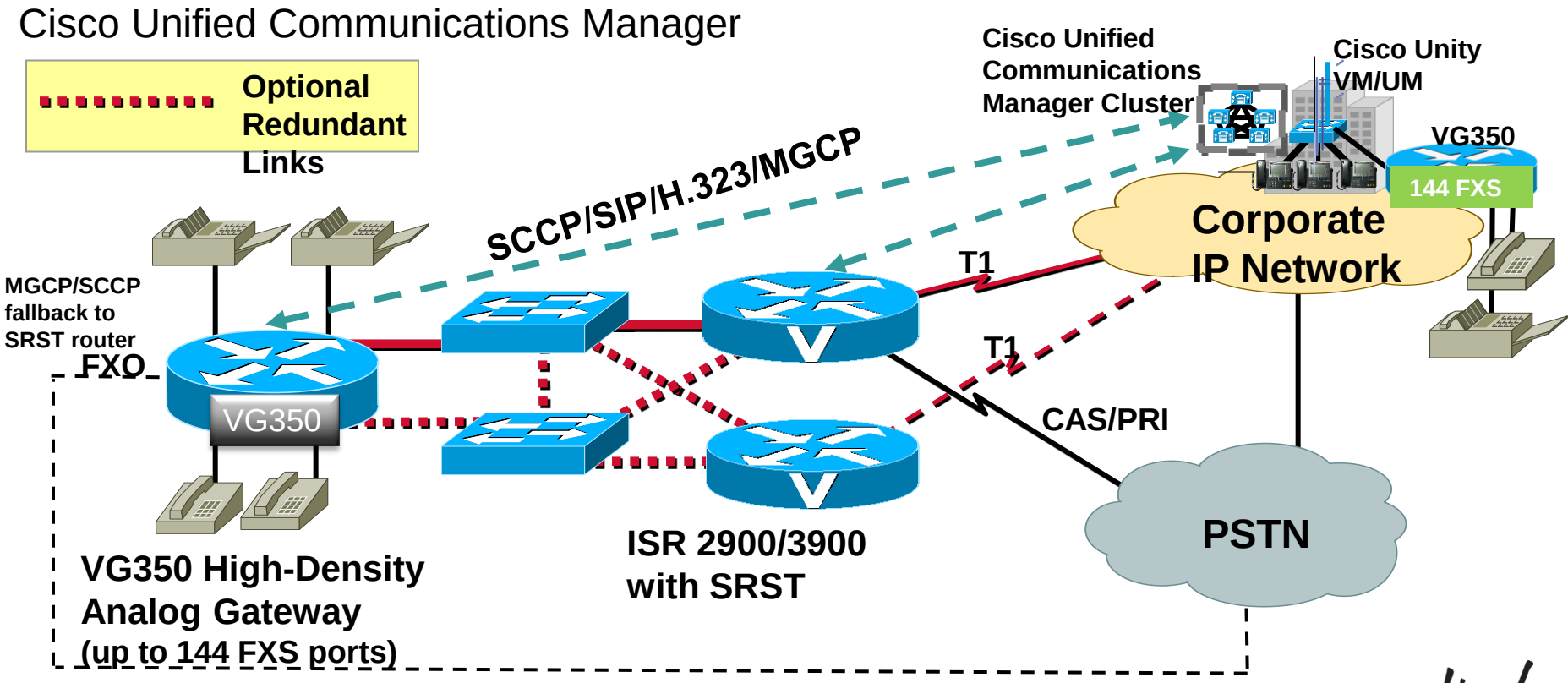
VG350 Deployment Topology



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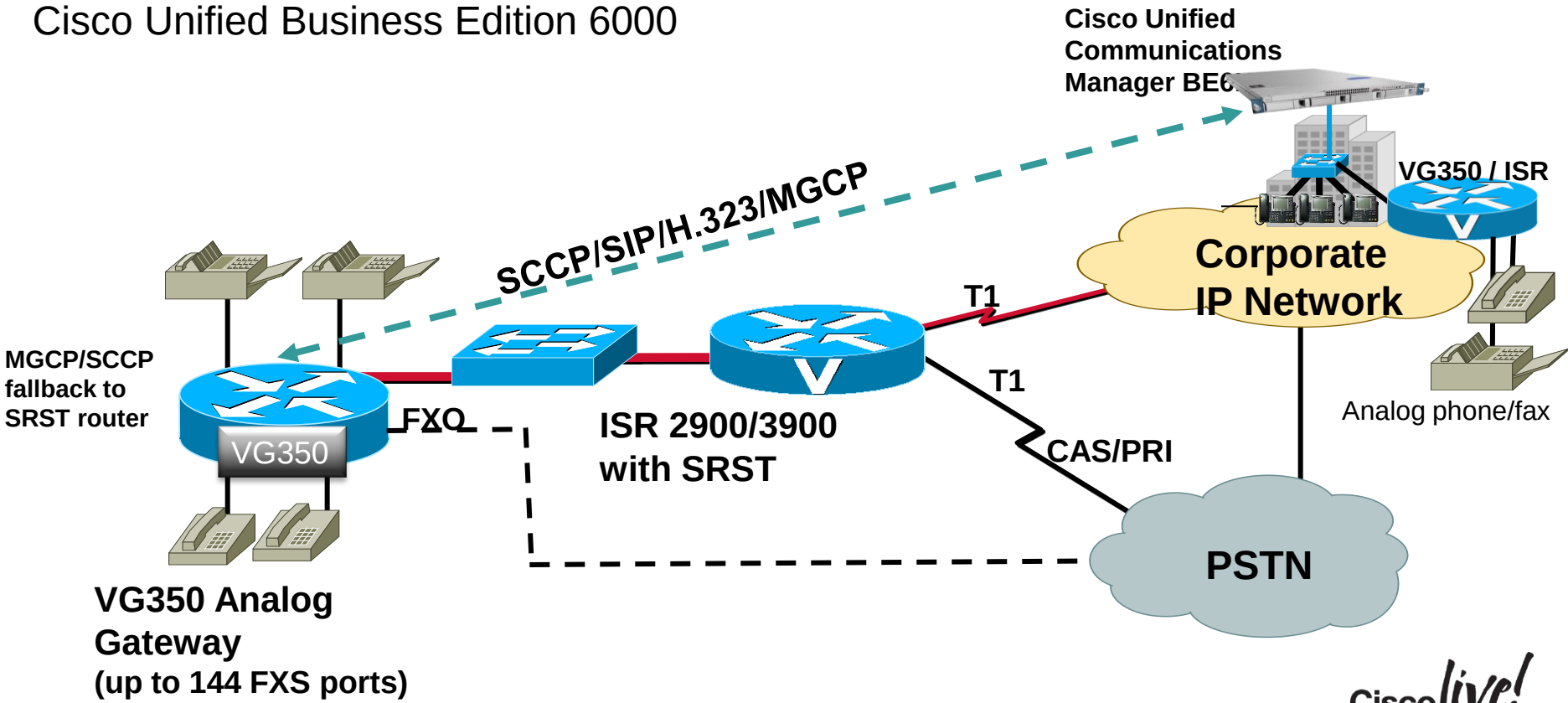
VG350 deployment topology

Cisco Unified Business Edition 6000



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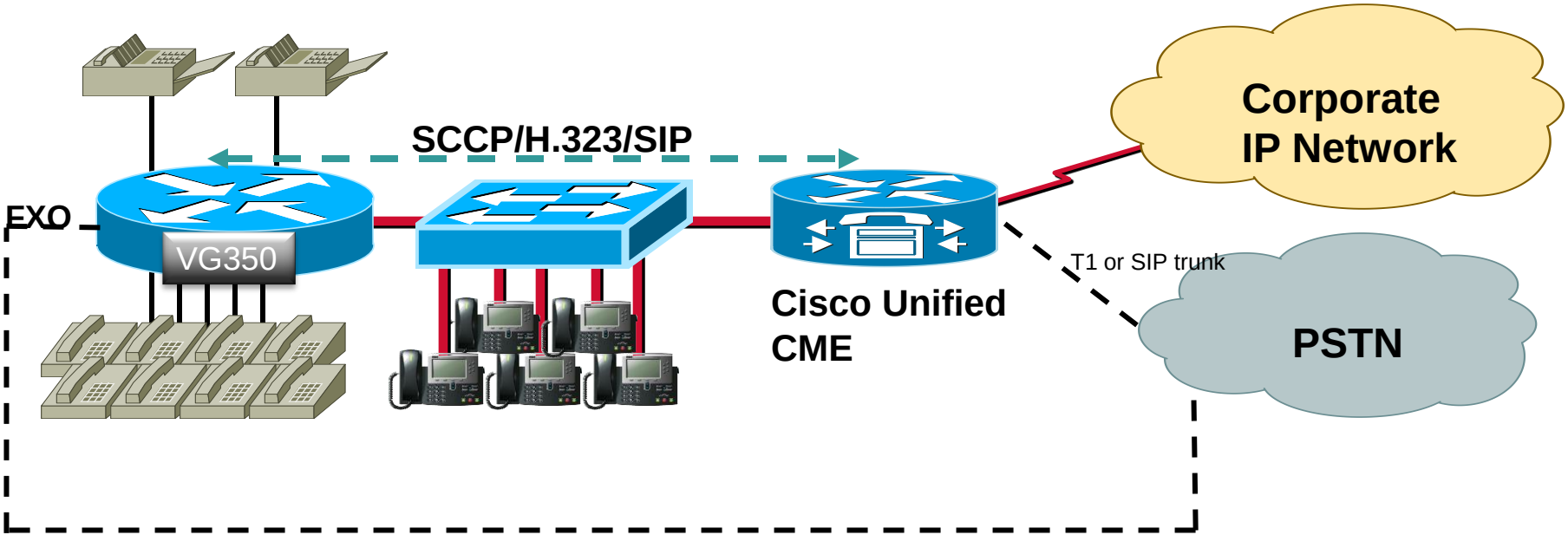
VG350 Deployment Topology
Cisco Unified Communications Manager Express



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Conclusion

- Cisco ISR platforms are best in-class voice gateways
- One platform supports TDM and SIP trunking enabling a simple transition path from TDM PSTN access to SIP trunk
- Cisco Unified CME provides distributed enterprise call control that includes business focus telephony features
- Cisco SRST Manager and E-SRST provide enhanced survivable UC end user experience during WAN outages
- Cisco UCS E-series delivers versatility and options on how to deliver UC at the branch
- Cisco VG analog gateways are best in-class voice gateways that deliver the benefits of VoIP to locations where analog end devices are preferred or required

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Cisco IWAN+Cisco UC concept

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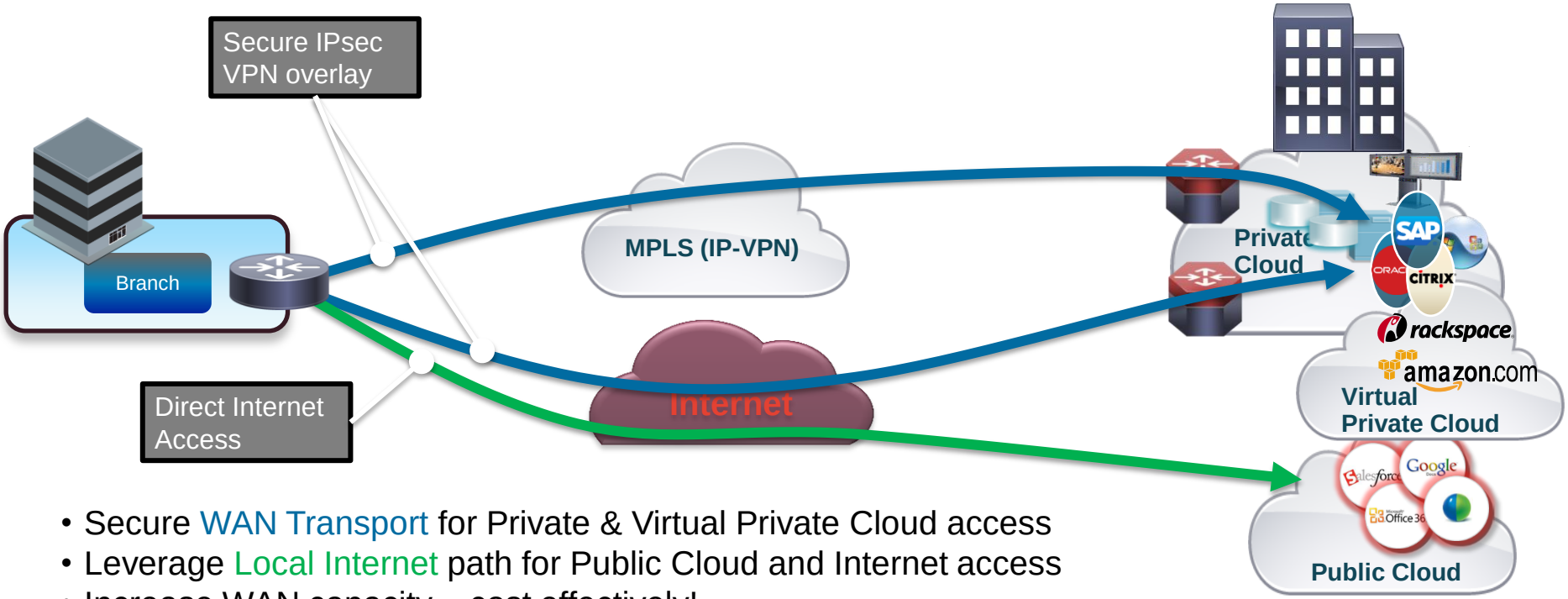
Intelligent WAN – Leveraging the Internet
Secure WAN transport & Internet Access



- Secure WAN Transport for Private & Virtual Private Cloud access
- Leverage Local Internet path for Public Cloud and Internet access
- Increase WAN capacity – cost effectively!
- Improve application performance (right flows to right places)

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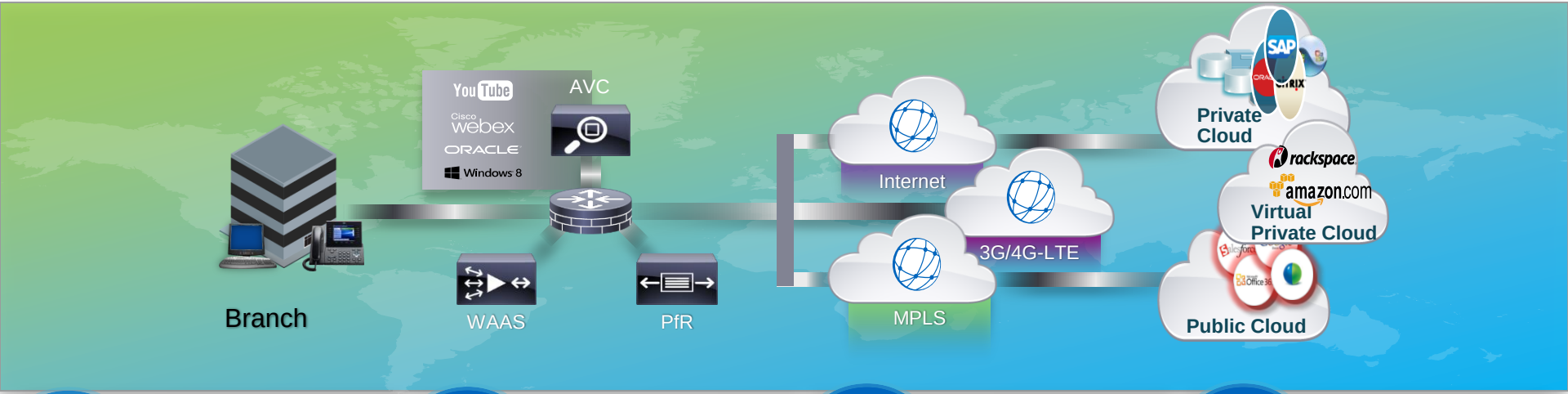
Intelligent WAN Solution Components



Transport Independent

- Consistent operational model
- Simple Provider migrations
- Scalable and Modular design
- **DMVPN** IPsec overlay design

Intelligent Path Control

- Application best path based on delay, loss, jitter, path preference
- Load Balancing for full utilization of all bandwidth
- Improved network availability
- **Performance Routing (PfR)**

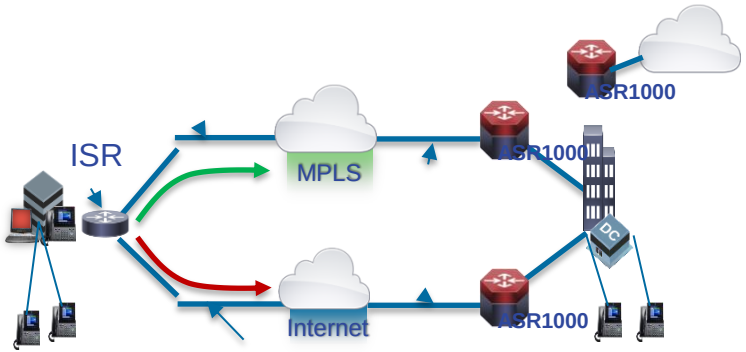
Application Optimization

- Application monitoring with **Application Visibility & Control (AVC)**
- Application Acceleration and bandwidth savings with **WAAS**

Secure Connectivity

- **Certified** strong encryption
- Comprehensive threat defense with **ASA & IOS Firewall/IPS**
- **Cloud Web Security (CWS)** for scalable secure direct Internet access

Audio and Video stream splitting using NBAR



- Voice and video conferencing traffic could be detected and given preferential treatment
- Video RTP streams are negotiated for a single RTP payload value, using the payload value NBAR can detect what kind of video is streaming and report that value to central controller (APIC-EM
- Example, training VODs server can use MPEG codec using payload 97, video conferencing can use G.711 codec payload 0 and H.264 codec payload 100.
- Using NBAR class-map detection the switch can then identify each stream detected
- PfR will route traffic based on qos policies set and determine if the traffic should flow across MPLS link or internet link

Note: voice traffic is always considered critical and would always be configured with the highest qos marking, forcing the traffic across the highest reliable link

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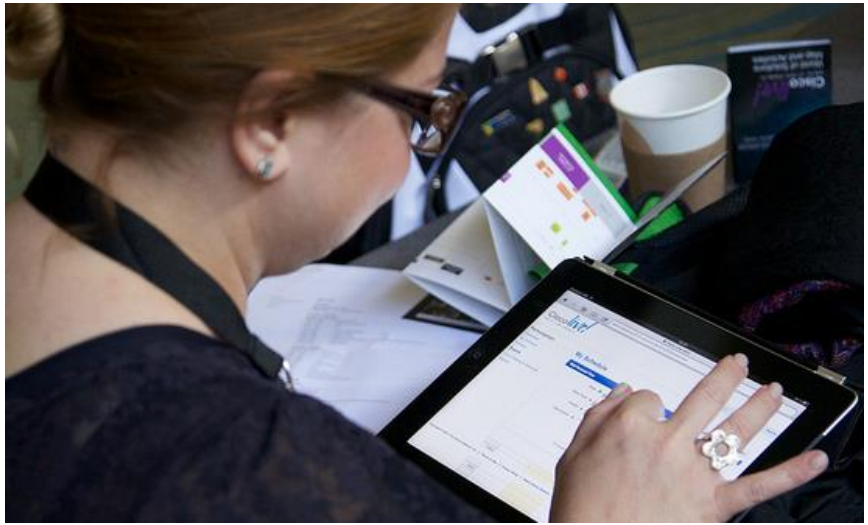

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